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Editorial

Standards? Who needs them?

Following the publication of the Final Report of the EBU/SMPTE Task Force for *Harmonized Standards for the Exchange of Programme Material as Bitstreams*, this issue of the **EBU Technical Review** includes further explanations of the work of this Task Force.

The EBU has long recognized that formal standardization is important for broadcasting systems and for production systems. Nevertheless, there are arguments in favour of *de facto* standards: for example, the audio CD gained rapid acceptance because it was demonstrably far superior to other products at that time. This meant that its inventors were able to avoid the long processes of consensus building in standards groups – in fact, formal standardization of the audio CD was pursued only after it had become successful in the market place. As some of the participants in standards bodies have a vested interest in preventing or delaying standardization of technologies favoured by their competitors, it is not surprising that some companies prefer *de facto* standardization.

EBU Members have long played a major role in setting standards. For example, the AES/EBU digital audio interface has been very successful. Similarly, in the early 1980s, the EBU and SMPTE collaborated very closely on the subject of standards for digital component video in the production environment. Throughout this work, the overt goal was to submit the outcome of the EBU/SMPTE's deliberations to the CCIR, which was acknowledged as the global standards body for matters related to broadcasting. At that time, standards (or "Recommendations" in CCIR terminology) had to be formally approved at the CCIR Plenary Assembly, which was held every 4 years. The EBU and the SMPTE realized that if they missed the 1982 Plenary Assembly of the CCIR, they would have to wait until 1986 for formal endorsement of their work. As such a delay was obviously unacceptable, there was great pressure to ensure agreement from the CCIR in 1982.

In retrospect, the CCIR's 4-year cycle was both an advantage and a disadvantage. The participants felt that they had to achieve agreement because they realized that it was "now" or "never". On the other hand, new developments might have to wait almost 4 years to be approved by the CCIR. It is only fair to note that ITU-R (the successor of CCIR) has successfully introduced procedures which allow more rapid approval of Recommendations.

Whilst recognizing the pre-eminence of the ITU, EBU Members now seem to place less emphasis on the ITU activities. Some of their standardization efforts has been diverted to voluntary groupings of specification providers, such as the DVB (Digital Video Broadcasting) Project or DAVIC (Digital Audio-Visual Council), or to collaborative research projects such as the Eureka 147 DAB Project.

The reasons for this drift away from the ITU are complex, but many EBU Members are disillusioned by the failure of the ITU to achieve world-wide agreements on standards. It is difficult to justify ITU Recommendations which recommend that users should adopt *"one of the systems, defined in Annexes A, B and C"*, where typically the Annexes refer respectively to systems developed in the USA, Europe and Japan. Although there is some

value in documenting such standards, the real goal of the standardization process should be to avoid multiple standards. In fact, it is unfair to place all of the blame on the ITU. The real culprits are the national and / or regional groupings who are too often driven by the *"not invented here"* syndrome. The ITU merely reflects the technical, commercial and political divisions of the real world.

The EBU realized in the early 1990s that it could not maintain its influence in the increasingly competitive world of standardization. Whereas RDS, NICAM and PDC were good examples of the EBU's informal role in standardizing broadcasting systems, it was clear that closer co-operation was necessary with the consumer electronics industry. It was also obvious that national standards acted as barriers to trade between the various countries of Europe and, furthermore, that European competitiveness was suffering because of the multiplicity of standards across Europe. To overcome these difficulties, the EBU joined with ETSI (European Telecommunications Standards Institute) to set up a Joint Technical Committee (JTC) on broadcasting standards. Although this venture was initially regarded with suspicion by some EBU Members, it is clear that the JTC has benefited both the EBU and ETSI, together with all sectors of industry – not just for broadcasters, but also for consumer electronics manufacturers, network operators and regulators.

The DVB Project has been very successful in obtaining agreement within the Project for its wide range of specifications. However, as the DVB Project is not a standards-setting body, it relies on ETSI and CENELEC to undertake the formal process of standardization. Some people have suggested that this additional step is hardly necessary in the case of DVB since many of the key players have participated in the DVB process. In practice, formal standardization brings some benefits, such as:

- enhancing the status of DVB specifications, notably allowing broadcasters to comply with an EU Directive which specifies that all digital TV services must conform to a standard adopted by a recognized European standards body;
- ambiguities or errors in specifications are occasionally identified in the process of public enquiry.

Almost every week brings details of a new forum aiming to set specifications for the converging worlds of telecommunications, broadcasting and computers. These fora aim to produce specifications very quickly, but this requires that all of the members share the same objective. Experience shows that conflicts of interest can occur even in such groups. If such disagreements do occur, some of the participants simply set up a new forum to promote their particular technology.

Given the bewildering multitudes of standards bodies, fora and specifications providers, the old joke *"The great thing about standards is that there are so many to choose from"* should, perhaps, be changed to *"The great thing about standards bodies is that there are so many to choose from"*.

Philip Laven

Philip Laven
Director
EBU Technical Department

Introduction

H. Schachlbauer
IRT

In this series of five articles, prominent members of the EBU/SMPTE Task Force describe the work carried out by the Task Force in pursuit of "Harmonized Standards for the Exchange of Programme Material as Bitstreams".

The articles have been derived from presentations given by the Authors at IBC '98 in September. Many thanks must go to Roger Miles and Myrianne Jansen of the EBU Technical Department for converting the audio tapes recorded at Amsterdam into word-processor text files.

The work of the EBU / SMPTE Task Force will have far-reaching consequences in the design of future television studios. Yet the subject matter of the Task Force's work is not really new: in fact it is a billion years old and is magnificently moulded into a painting by Michael Angelo Buonarroti (see *Fig. 1*) which contains all the essential elements that we are actually dealing with.

Of course when the Lord decided to create a new system all those years ago, He had to think about the essential elements that would be required. As you can see from *Fig 1*, He and his staff arrived by means of a *streaming transfer*, so at least He knew about streaming. He delivered the draft version of the operating instructions in the form of "ten commandments", using a file format whose closing code is "amen". He was also very aware that whenever you assemble things in a *wrapper* where you keep everything together, some people will suffer from *compression*. Furthermore, his draft operating instructions already asked the basic question *is this really **metadata** or is it **data essence***? – this is something we are desperately trying to sort out today.

In the painting you can also see that the Lord is trying to "ping" an object and is attempting to establish a *protocol* for the liaison. You can further see in this painting that He has given very deep thought to the need for an *open interface* – for which we are very grateful as we are basically concerned with the same issue today, but of course with a different reach!

The work of the Task Force has been concerned with the collision of two worlds: the world of *Television* and the the world of *Information Technology* (IT), both of which operate on very different paradigms.

In the TV world, we have been heavily involved for some while in the process of *digitizing* – mainly trying to improve our assets in terms of maintaining the integrity and quality of the audio and video, and maintaining an adequate post-production headroom to carry a coded signal through a long and lengthy post-processing process. This has all been set in concrete by a range of standards from the EBU, SMPTE, ISO, ANSI, ETSI and ITU-R. As broadcasters, we have really taken digital technology on-board – but with almost the sole

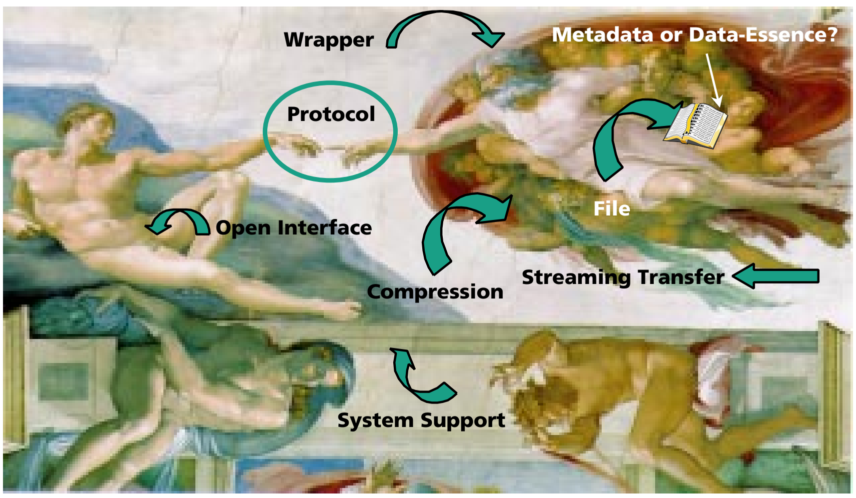

Figure 1
The first call for Open Standards and agreed Interfaces.

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focus of maintaining the quality of our products (of course by maintaining the quality, we are also opening new horizons for those people who want to do something creative with their content).

There is now another world coming towards us and there is great risk of an imminent collision. We certainly do not want it to collide with our TV world; rather we want to have a "controlled crash" – as pilots describe the landing of an aircraft.

The IT world has very different rules. Nothing (or very little) is set in concrete, there are different operating platforms (not really interoperable) and few things are palpable, although some work is going on in the ITU-T in terms of standard interconnections. The great plus of the IT world is that it provides *functionality* that we must take on-board. In the future, it is most likely that we will have to run our TV operations in a fully-networked environment, where it is not only a question of maintaining the quality but also a question of:

- improving the functionality of our systems;
- maintaining our assets;
- getting right the administration of our assets;
- opening new business models to make money.

There are also a number of elements which are common to both worlds (see *Fig. 2*); for example, MPEG, ATM, DV and Motion JPEG. These all need to be carefully defined and settled to really make things click. This was realized two or three years ago when the EBU and the SMPTE set up working groups within their respective bodies to discuss these matters, but never got very far with them.

We soon realized that we would have to assemble a really critical mass at an international level in order to obtain a high public profile and to get the manufacturers and users on our side. In that way, we could determine the path and course of action needed to ensure that this new technology merger ended up not in chaos but in something useful, efficient and economic. Thus, the EBU and the SMPTE [1] got together to define a number of items that needed standardization, or at least needed some directions or recommended practices in order to make a whole networked studio environment become a

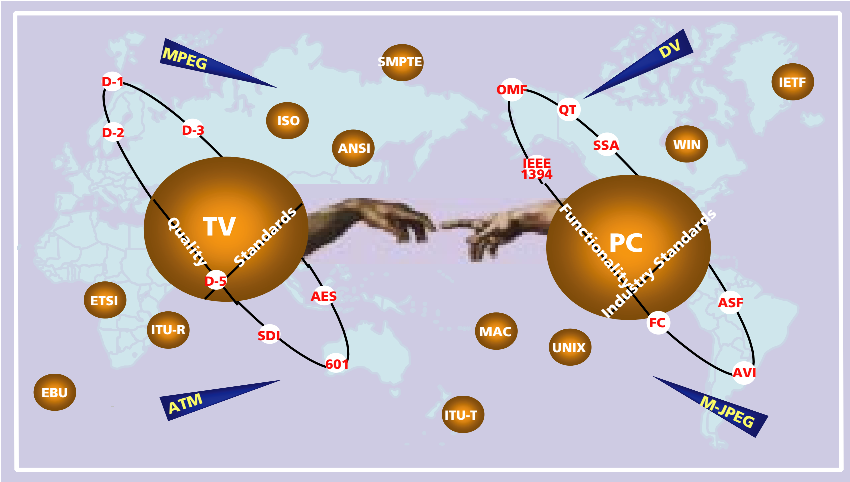

Figure 2
The TV and IT worlds on a collision course.

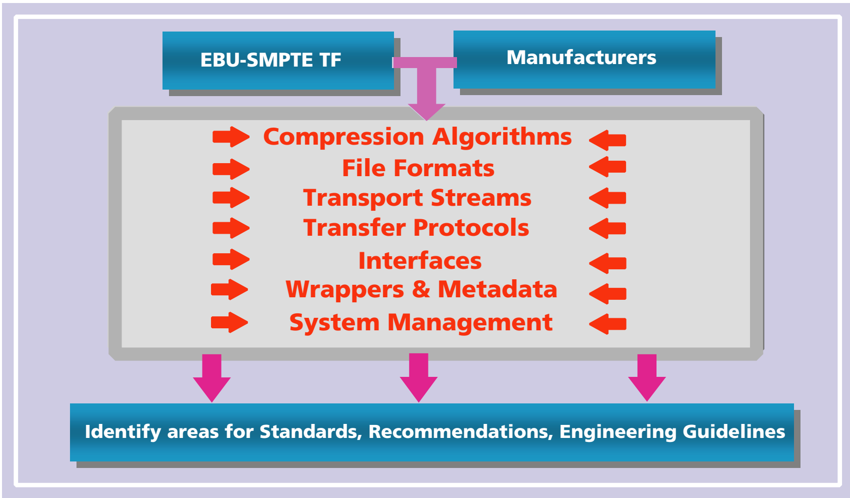

Figure 3
Areas identified for Standards, Recommendations and Engineering Guidelines.

real possibility. These discussions resulted in the creation of the EBU / SMPTE *Task Force for Harmonized Standards for the Exchange of Programme Material as Bitstreams* which went on to identify a number of items of great concern to the TV world (see *Fig. 3*).

Compression algorithms were one such concern as they were already imminent. It was thus a question of looking at the issue and seeing what we could do in terms of directing future activities. The whole issue of *file formats* is not new to television of course. In some areas such as graphics, we've been using file transfers for some time, but not in a way that it could be spread ubiquitously over the whole studio environment. New *transport streams* and *transport protocols* would have to be defined, and *interfaces* would have to be defined to interconnect signals seamlessly from A to B with equipment from different manufacturers at either end. Of course there was this new thing called *Wrappers & Metadata*.

Wrappers are not very familiar – in Europe they used to be called "containers". A wrapper is where you put something in, but we are not wrapping just the remains. Rather, it is the valuable assets that we are wrapping. Metadata is mainly the administrative

1. The EBU purely as a user group; the SMPTE also as a user group but with very strong connections with the manufacturers who use the SMPTE as a platform to define standards for themselves, in the knowledge that it is important to have standardized products if they are to operate successfully in the marketplace.

information that we need to select material if it is going to be used in a multiplicity of different outputs. Of course the most difficult thing of all, and we've unfortunately realized this very late in the day, is to tie all these things together and to mould them into a system approach which does not require specific skills from an operator. Rather, it should have an inconspicuous layer – *System Management* – which does it all and leaves more freedom to those creative people who want to use the system in a creative way.

We started the Task Force activity at IBC in Amsterdam two years ago (1996) at a press conference where we set the goals. We were very ambitious in thinking that we could settle everything by the next NAB conference (1997) in Las Vegas (see *Fig. 4*). That was an illusion – we painfully became aware of this as we approached the date of the NAB conference.

The only thing that we could provide was a first report of a tutorial nature,

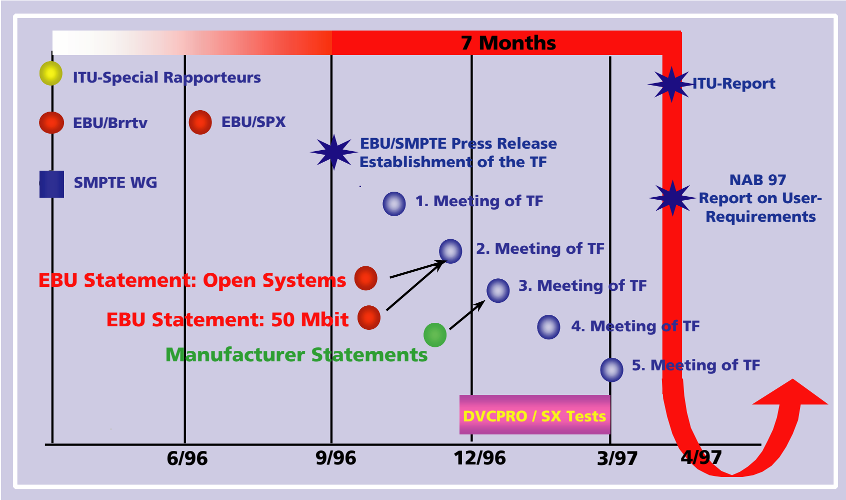

Figure 4
Achievements during the first round.

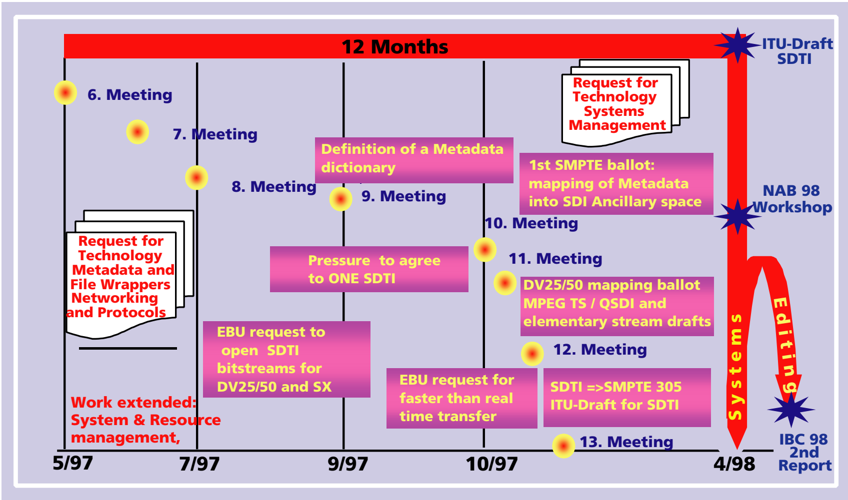

Figure 5
Achievements during the second round.

where we analyzed what the issues actually were and presented some user requirements that we had assembled, but nothing else. That had a value in itself, but it was not so very productive because it did not tell people where they had to go. It just stated what a mess there was and what we actually wanted, but it did not provide a clear vision of the direction where manufacturers and users would have to go. So, we had to carry this overspill into a second round which was much more productive.

In this second round, we set about issuing requests for technology to the outer world where we knew there were already a good number of solutions. We just had to pick the right ones which could be adapted appropriately to the specific needs of the television community. These specific needs were mainly in the areas of (i) gross data rates, which you normally don't encounter in office applications, (ii) the question of latency and (iii) the question of deterministic control of all these things. So we had a range of requests for technology coming and going between the Task Force and the proponents of the various technologies. The responses were duly analyzed and we came to some conclusions.

Fig. 5 shows the milestones where we reached some agreement with the manufacturers over things needing to be done to make things work. One of the key user requirements was that any system accepted for use in a broadcast environment would have to be laid

Abbreviations

ANSI	Americal National Standards Institute
ATM	Asynchronous transfer mode
ETSI	European Telecommunication Standards Institute
IBC	International Broadcasting Convention
IEC	International Electrotechnical Commission
ISO	International Organization for Standardization
ITU-R	International Telecommunication Union, Radiocommunication Sector
ITU-T	International Telecommunication Union, Telecommunication Standardization Sector
JPEG	(ISO/IEC) Joint Photographic Experts Group
MPEG	(ISO/IEC) Moving Picture Experts Group
NAB	National Association of Broadcasters (USA)
SMPTE	(US) Society of Motion Picture and Television Engineers

open in all its properties and it would have to be standardized officially within a body such as the EBU, the SMPTE or any other recognized standards body; it could not be proprietary. A second issue was to generate quality guidelines for manufacturers on how we think that compression should evolve.

The Task Force's work was a truely collaborative venture between the users and manufacturers. *Fig. 6* shows a list, which is not exhaustive, of the major players in Europe and North America who participated in the work. We all realized, the users and the manufacturers, that we were working to achieve the same goals. As users, we knew that we would only install new systems on the premise that all the components would be interoperable, even if these components were installed on a piecemeal basis. We were not going to promote any "turnkey" solutions.

We knew that we would have to guide broadcasters from different starting points carefully into this future environment which we can only vaguely see; the technology is moving so fast that we don't have a total vision of where we are going to end up.

To sum up, the Task Force's work was a collaborative effort involving more than 200 people, costing an estimated 2,000,000 US dollars. It has taken that much to achieve what is now printed in two books – the Final Report of the Task Force which has simultaneously been published by the EBU and the SMPTE. *(The EBU version was distributed as a Special Supplement to the Summer 1998 issue of EBU Technical Review.)*

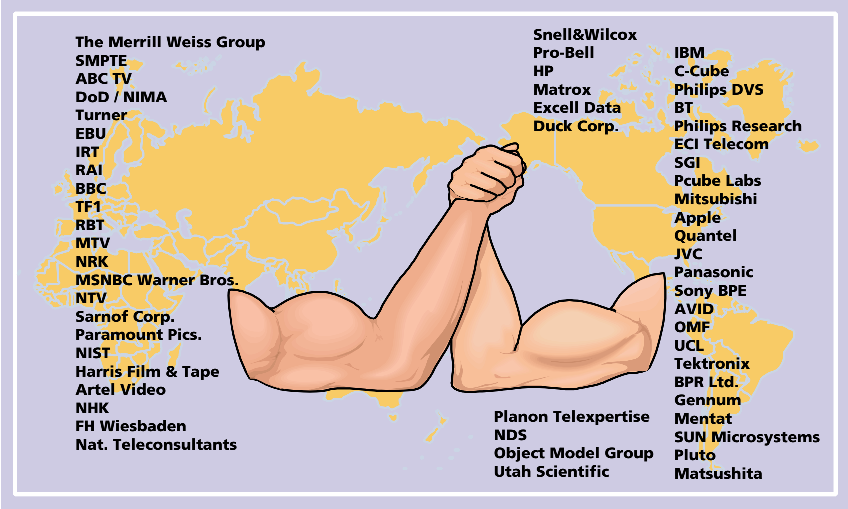

Figure 6
List of members of the EBU/SMPTE Task Force.

Horst Schachlbauer *graduated in Telecommunications from the University of Munich in 1967 and has since worked for the IRT, the central Research Laboratory of German, Austrian and Swiss public broadcasters.*

Mr Schachlbauer been very involved in the development of standards for digital television on national as well as international platforms, e.g. ITG, EBU, CCIR and ETSI. In particular he was involved in the specification of CCIR Rec. 601, the D-1 recording format, the ***S****erial* ***D****igital* ***I****nterface and PALplus.*

Currently, Horst Schachlbauer heads the EBU MAGNUM committee which closely liaises with manufacturers in the area of recording technology for television. He also chairs a number of national and international Project Groups dealing with digital television production technology and archiving. He served as the European co-chairman of the EBU/SMPTE Joint Task Force and has recently been elected a Fellow of the SMPTE.

The work of the Task Force is not finished yet. There is still a great deal of standardization work to be carried out and this will be done mainly through the SMPTE which has initiated a new organizational structure for its standards development activities, in order to be reflective of the Task Force's work.

A PDF version of the Final Report of the EBU/SMPTE Task Force can be downloaded from the EBU Website at:

http://www.ebu.ch/pmc_es_tf.html

. . . or from the SMPTE Website at:

http://www.smpte.org/engr/tfrpt2w6.pdf

But be warned, this is a LARGE FILE (2.7 MB).

Systems

S.M. Weiss
The Merrill Weiss Group

Future television production systems are going to be far more complex than anything we've dealt with in the past. Thus, in order to pull together all the various dimensions of what the EBU/SMPTE Task Force had set out to do, it created the Systems subgroup specifically to concentrate on matters relating to system integration.

This article describes a system model for television production operations, based on the transfer of bitstreams within a distributed studio object environment. It puts forward a migration strategy and also describes a different economic model for equipping television production facilities in the future.

1. Introduction

Traditionally, broadcasting operations have been based on a single distribution channel that produces an integrated continuous programme as its output. More recently, we've seen a proliferation of distribution mechanisms to the consumers / viewers, and there is today much discussion about the need to "re-purpose" programme content for use across these newer distribution channels.

Arising out of this diversity, future television production systems will be quite different in the way they are structured when compared with today's systems. Traditionally, systems have generally been built around tape recorders and linear equipment which run in real time. Systems of the future, on the other hand, will use servers and will be able to move programme items around much faster than real time, or much slower than real time, to take advantage of bandwidth-versus-cost trade-offs that are becoming possible in the new digital world. New workflows will result, and we will have the capability, for instance, of having a series of different processing capabilities located at different facilities, and having those facilities interconnected on an ad-hoc basis under the control of a single operator. This will be achieved by using *object-modelling* techniques. By using the interconnectivity that will become available, we will be able to take a system that is distributed and have it appear on a single monitor screen, thus giving an operator the chance to concentrate on the functionality of the content and not have to worry about the technical complexity that really exists underneath the whole system.

In the past, when we designed a system, we were concerned about what would be connected to what – the hard wiring of the system more or less determined the functionality that the system could provide. Thus, it was important to keep accurate documentation on which cable went from where to where, in order to find a connection quickly for maintenance or fault-correction purposes. In the future, we will be more concerned about where we need to locate the servers and the various kinds of processors that make up the system. We will also be just as concerned about selecting the right drivers and the right software to load onto those new systems. We will also need accurate documentation if we are to control and maintain the software, its various revisions and its various configuration files – all of which may be spread across different parts of the system.

2. Object models

In order to plan the control and representation of a system, we decided fairly early on in the systems review process to work with *object models*. For those who are not familiar with object models, there is a good description of how they work in Annex B to the Final Report of the Task Force. Basically, we have selected object models both for the control of systems and also for the representation of content.

One thing that an object model does is to integrate the treatment of the processes and the information. Thus we can have a single representation or a single structure that can deal with, for instance, the functionality of a server

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and the content that is on that server. By using object models we are able to divide the management of those things into blocks which can be further sub-divided. Then, for instance, if we move some programme material from one place to another, we can have the associated metadata transferred with it, if that is appropriate.

Object models also allow us to have updates to the software, and to control the functionality in confined areas of the system without causing us then to have to make changes in other areas at the same time. They also permit us to have extensions to functionality through the use of registries. Ultimately we may get to a point where the appropriate drivers come with the purchased equipment and can be loaded onto any other device that needs to control that equipment – without needing to have a new interface written by every manufacturer. For example, whenever a new tape recorder comes on the market, the manufacturer will supply a driver that can be installed on all the existing editing systems. This flexibility arises from a recognition by many control system manufacturers that there really is no added value for them in writing drivers – it is simply a cost and a drain on resources. If we can set things up in such a way that it only has to be done once instead of being done by everybody, then there are significant benefits both for the manufacturers and the users.

Abbreviations

API	Application programming interface
FTP	File transfer protocol
GPS	Global positioning system
IEC	International Electrotechnical Commission
ISO	International Organization for Standardization
MPEG	(ISO/IEC) Moving Picture Experts Group
OSI	Open systems interconnection
SDI	Serial digital interface
SDTI	Serial data transport interface
SMPTE	(US) Society of Motion Picture and Television Engineers

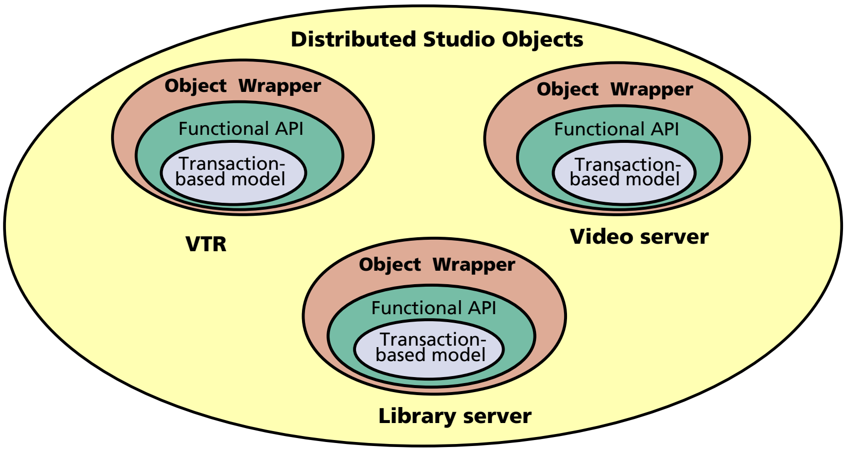

Figure 1
A studio based on distributed objects.

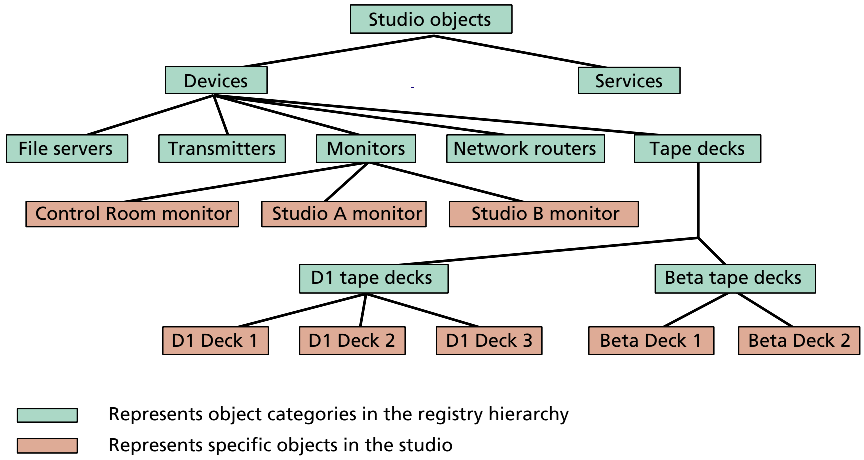

Figure 2
Network object registry.

2.1. Distributed objects

Fig 1 illustrates a studio which consists of distributed objects. Inside the small circles are *transaction-based models* – which refer to the kind of control that we have had in the past (where we probably had RS-422 networks running at 38.4 kbit / s, over which could be sent a command that said "play" and another command that said "stop," and so on).

Wrapped around these models are a more-recent phenomenon which has come out of the IT world – *functional APIs* that allow us to make various kinds of calls to devices and to control them on a much more software-oriented basis. Surrounding the functional APIs are *object wrappers* that allow us to have a single approach to a given device type. Thus, in the case of a tape recorder (VTR), if we approach the object wrapper and say "play," it will understand our command and translate it to any chosen machine within the wrapper. By having wrappers appear at different places throughout the system it becomes distributed, and that is why we call these *distributed studio objects.*

2.2. Object registry

Ultimately, to make all of this work, we will have to have a *network object registry.* A sample of the hierarchy of such a registry is shown in *Fig. 2*. *Object categories* in the diagram are shown in green, while *specific objects* are shown in brown. At the top of the hierarchy, we have *studio objects* and below that we have *devices* and *services.* Under devices we have such things as *file servers, transmitters, monitors, network rout-*

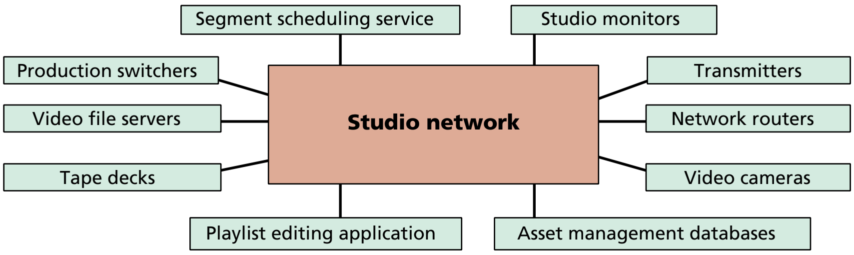

Figure 3
Concept of a network studio.

ers and *tape decks*. In the case of monitors, just as an example, we have a *Control Room monitor*, a *Studio A monitor* and a *Studio B monitor*. In the case of tape decks, you can see that there are three D1-type tape decks and two Beta-type tape decks in this particular system.

All this equipment would appear in a central registry which would allow us to control them from anywhere on the network, to learn about their capabilities and to make use of them. Another way to look at this arrangement is in terms of a *networked studio* where we have a studio network that views, equally, those objects that represent physical devices and those objects that represent software entities (*Fig. 3*). They are all part of the network and are perceived equally, and that is a major reason why object-modelling techniques work very well for studio applications.

3. System model

Over the long term, if there is one diagram that will come to represent the work of the Task Force, it is probably the system model (*Fig. 4*). Some of the other Task Force subgroups may well have their own favourites, but certainly within the systems area, this is probably the one diagram that tells the biggest part of the story.

Fundamentally, in television production, we have a number of *activities* which are represented on the model by the blocks distributed across the first vertical plane (*Video Essence*). Notice that there are equivalent activity blocks on each of the other vertical planes behind the first one, i.e. the *Audio Essence*, *Data Essence* and *Metadata* planes. Cutting through all the activities attached to the vertical planes are four horizontal *Communication Layers*, namely *Applications*, *Network*, *Data Link* and *Physical*. And underlying these two sets of orthogonal planes and tied to all of them is a *Control and Monitoring* plane.

3.1. Activities

Activities are basically the various stages of "production" that we are familiar with; for instance, we start with pre-production and move on to acquisition and production, then on to post-production, distribution, storage, transmission and archiving. We can take those seven activities and use them to model virtually anything that we wish, in terms of television production. Of course, these definitions will be specifically different if, for example. we are referring to news rather than a situation comedy, but in general they will be applicable to all instances of production.

3.2. Essence and metadata

The next thing we want to look at on the model are the vertical planes – there are four of them. The first is *Video Essence*. If you have read the first Task Force report (April 1997), you will remember that *Essence* and *Metadata* together are equal to *Content*. In the terminology of the Task Force, when we talk about video essence, we are referring to streams of video only. Similarly, *Audio Essence* refers to audio streams only. Video and audio files, on the other hand, are treated as *Data Essence*.

Data essence is information that inherently has stand-alone value; it is not video essence or audio essence. Thus, for example, teletext and closed captioning (a real-time subtitling system used mainly with analogue TV systems in 525-line countries) are considered to be data essence as they both have meaning on their own.

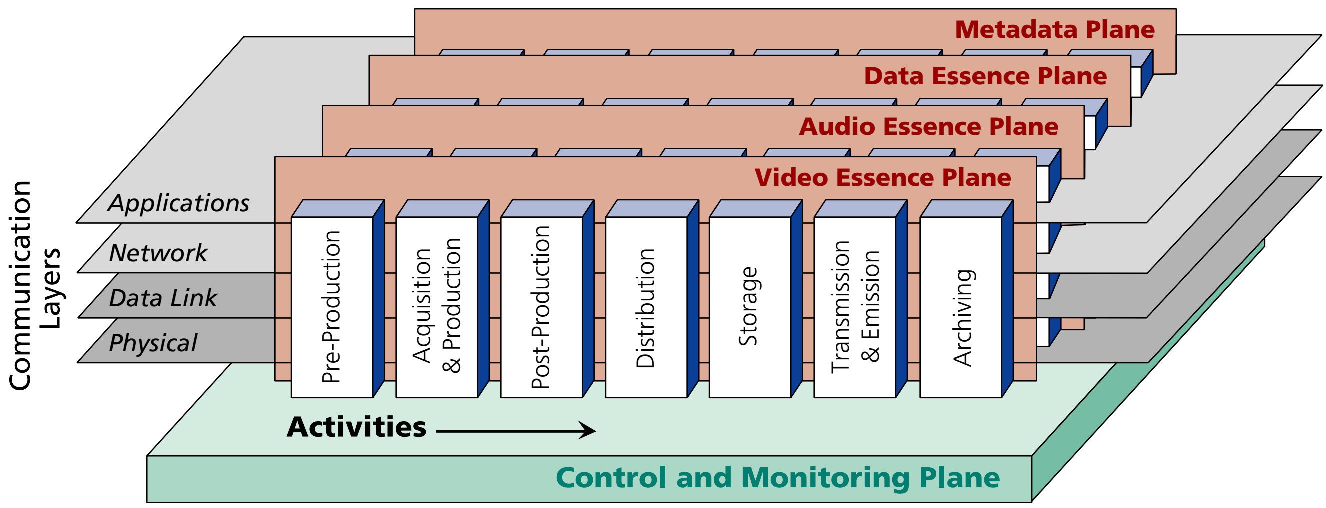

Figure 4
The system model comprising activities, planes and layers.

Finally we have metadata, which has no stand-alone value. It is tied to the video, audio and data essence in some manner. A good example of metadata is timecode: when taken by itself, it has no meaning but, in association with a frame of video essence, it allows that frame to be identified.

3.3. Communication layers

Looking again at our system model in *Fig. 4*, we have various communication layers that are represented by the horizontal planes which transect the model. These communication layers are in many ways similar to the familiar ISO/OSI seven-layer stack but, in our case, we have decided that there should be just four layers to represent a television facility, or a television system. We start with the *Physical* layer on the bottom which carries the electrical and mechanical characteristics of the system, or the interconnection. We then have a *Data Link* layer that involves the protocols that interconnect the *directly-connected* devices. So if a controller is connected to a VTR, the data link layer controls the direct communication between those two items. The *Network* layer above that controls the protocols between *indirectly-connected* devices. Finally, on the top, we have the *Application* layer that deals with both specific applications and what the ISO/OSI model would call the presentation layer.

3.4. Control and Monitoring plane

Underlying the aforementioned elements of the model, we have the *Control and Monitoring* plane. In most respects, it touches all the other elements of the model, basically providing co-ordination between everything else that sits above it. In the case of transfers, storage, manipulation, monitoring, diagnostics and so on, the control and monitoring layer also provides overall management of content across the various activities, planes and layers. Thus the control and monitoring plane not only performs a control function, it also provides a management function. For example, if we need to allocate bandwidth between devices, this matter will be handled by the control and monitoring plane.

Ultimately, this plane provides the interface to the human operators of the system.

4. Television operations

The next thing to consider is what we can do with the system model, in terms of television operations.

There are various aspects of operations that are of concern to us:

- control;
- monitoring, diagnostics and fault tolerance (which, when taken together, are related to control but which otherwise are separate from it);
- data essence and metadata management;
- content multiplexing;
- multiplexing essence into containers;
- timing, synchronization and spatial alignment.

4.1. Control

Control can be split into a number of types; for example, there is *strategic control* which is at the fundamental planning level; *tactical control* which determines how we are going to make things happen, and *peer-to-peer control* where one device sends information to a second device in order to make control possible and to achieve the tasks that the second device has been designed to do.

Managing resources is another type of control. For example, the control of bandwidth or data space in a server, or what you are going to do if you have only a certain time availability on equipment, are all resource management issues.

Yet another type of control is the physical control of networks. If equipment or subsystems are to be connected together, we have to make sure that the control system understands what is linked to what, what can connect to what and, if something cannot be directly connected to something else, how we are going to send our content there. For example, if our content was recorded on a DV tape, but we are using an MPEG-based editing system, the control system has to know that when the DV content is sent to the editing system, it must be passed through some sort of a translating device on the way – so that when it arrives at the editing system, it is in a compatible format.

There are various forms of control implementation, as already discussed above in terms of object models. The Task Force will be implementing control through the use of various logical control layers, and that ties in with some of the layering already mentioned above.

4.2. Monitoring, diagnostics and fault tolerance

When we think about monitoring and diagnostics, we are really referring to a feedback process that allows us to get information about what has happened and to feed this back to a controller. We must be able to predict failures by monitoring our systems and knowing how they are performing. We must also be able to carry out diagnostics on-line, while the system is in operation, and off-line by taking part of the system out of service.

Redundancy has typically been used in broadcasting facilities to provide *fault tolerance* in systems. However, fault tolerance has also been achieved through the segmentation of systems so that if there is a failure in one segment of the system, it does not have to affect the entire system. Fault tolerance in systems has also been achieved through various monitoring and verification processes.

4.3. Data essence and metadata management

When we think about data essence and metadata management, again we are referring to the layered structure of our systems model. We have a number of ways of handling content transfers including:

- file transfer mechanisms;
- streaming.

Streaming is fundamentally an isochronous process which allows us to send content at a continuous rate from one place to another. File transfer, on

the other hand, allows us to re-send the content if there has been a problem. They are very similar to the typical file transfer that occurs when downloading from an FTP server on a computer network.

It must be possible to link together data essence and metadata when necessary, but to keep the metadata in a separate place so that it can be treated within a searchable database. We also have to be concerned about whether the metadata that relates to particular essence is permanent or temporary. For example, when we send a control command as metadata along with essence to a particular device, that command only has a life from the time it is issued until the time it is executed. There are other metadata elements that may have much longer lives and again others that may need to be changed as they pass along the production process. We have to be able to recognize what the lifetime of a metadata element should be, and to properly maintain it until the end of its useful life.

4.4. Content multiplexing

Next we come to content multiplexing. Here we can have either a single-step multiplex or we can have cascaded stages. Several of the possibilities are as follows.

- We could have individual channels that have to be put together to form a multiple-channel component. An example would be multiple audio channels that multiplex together either to form a stereo pair or a 5.1 format surround sound audio signal.
- We might assemble an SDI stream or, similarly, an SDTI stream from multiple components.
- We might form a content package from multiple individual components.
- We might form a multi-programme package from several different existing single-content packages.
- We could take elements that are already multiplexed into various packages, strip them out of those packages and put them together in yet other packages – this is often called *grooming.*
- We might assemble multiplexes for faster-than-real-time transfer, or for slower-than-real-time transfer.

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Mr Weiss has been involved in the development of standards for the television industry for over 21 years. He organized and produced the tests that led to CCIR Rec. 601, the foundation for all digital television standards since. He has contributed to many other standards in areas including digital control, the Serial Digital Interface (SDI), the Serial Data Transport Interface (SDTI) and Video Index, a precursor to the more generalized form of Metadata.

Currently, Merrill Weiss serves the SMPTE as Engineering Director for Television, in which role he is responsible for organizing all of the SMPTE's television standards development activities, and he chairs the SMPTE Television Steering Committee. He is a Fellow of the SMPTE and a recipient of the David Sarnoff Gold Medal Award.

Mr Weiss served as the SMPTE co-chairman of the EBU/SMPTE Joint Task Force and also chaired the Systems subgroup.

4.5. Multiplexing of essence into containers

We have to be able to transfer content between multiplexes. If some content comes in as part of a multiplex and needs to go out as part of a different one, we must be able to handle that – again, this is in the domain of "grooming."

If we have a certain amount of content that we are trying to convey across a channel, but if that content does not fully fill the channel, then we may want to send additional data along with the content in order to fill the channel – this is called *opportunistic data*. We have to be able to control this use of the channel, and there is already some standards work going on to support such applications.

We also need to be concerned about the multiplexing of signals originating in different systems and formats, e.g. DV compression and MPEG compression. So how do we put this type of multiplex together and convey it from one place to another? We would not wish to have separate infrastructures to handle different formats or systems. The Serial Data Transport Interface (SDTI) is one very important solution to this problem, the completion of which was due largely to the efforts of the Task Force.

Finally, in this section, we must consider statistical multiplexing. If we have a number of things we wish to convey through a channel and we want to be able to share that channel in the most efficient way, then we will want to allocate the bandwidth dynamically to each of the encoded signals – without in any way compromising any of the respective content elements.

4.6. Timing, synchronization and spatial alignment

When we consider putting all these elements together into a system, we have to start thinking about issues of timing and synchronization in rather different ways from how we thought about them in the analogue days. Of course, we will still need reference signals. Up to now, we've used the black-and-burst reference signal of analogue systems. The thinking until fairly recently was that the same black-and-burst would even do for the digital world, but that thinking is now changing somewhat. There will probably need to be an augmentation of the information that can be carried in a black-and-burst signal to enable us to achieve some additional timing and sychronization objectives.

We are also likely to require some form of absolute time reference – something that has not been needed in the past. This, at least in the thinking of the Task Force and in other interested bodies, is likely to be a GPS-based reference signal.

In analogue systems, we have been able to use frame synchronizers to

achieve temporal alignment. When the clocks at the two ends of a circuit were running at slightly different frequencies, a buffer placed in the middle could overflow or underflow and, by simply repeating or skipping a field or a frame, the problem could be fixed, albeit with a little hiccup in the signal if somebody was watching very closely.

In the compressed digital domain, if we were to use a buffer to absorb timing differences – for instance those caused by Döppler shift in satellite links – we would have a much more serious problem if the buffer were to overflow or underflow. There is no simple way to jump forward or backward in small increments, and the irregularity of the data repetition exacerbates the matter. However, the application of an external frequency reference to all the systems in a network will keep them from drifting apart, eliminating the possibility of buffer overflow and underflow, so long as the buffers are large enough to absorb the variations that occur in the network delay.

Temporal alignment of compressed signals as they pass through concatenated compression processes is an important consideration if quality is to be maximized. This means ,basically, that we must code a given frame in the same way each time – so that an I-frame remains an I-frame, for instance. With the many compression schemes that are likely to be cascaded, it may be impossible to maintain temporal alignment rigidly, but it should be managed to the best extent that other considerations allow.

When compressed signals are to be processed in certain ways, it may be desirable to constrain the compression systems in ways that optimize the efficiency through a system, even though the compression process itself may operate at less than maximum efficiency. Thus it may be best to use constant values for the number of bytes and the number of frames in a GoP when the primary objective of a process is editing. This will reduce compression efficiency by wasting the bandwidth but may yield improved overall efficiency.

Latency is the delay that occurs between the input to a process and its output. Compression, by its nature, has associated latency. The greater the amount of compression, the higher the latency is likely to be. When playing back recorded content, latency can be compensated for by back-timing the start of the playback. When live programming is compressed and interactions are required between participants in different locations, latency can become a significant complicating factor. Special attention is required when using such techniques as audio clean feeds.

When compression systems are concatenated, spatial alignment of the image – so that block and macroblock boundaries always fall in the same place – is an important consideration in maximizing the image quality. This can be very difficult to achieve in an environment of mixed compression schemes. Within a single family, it should be achievable and is recommended.

During a transition period that is likely to be somewhat drawn out, both analogue and digital signals will be in use in hybrid facilities. In such operations, account must be taken of the time required to convert between signal forms; the video, audio and data must remain time-aligned despite passing through different processes. It will be necessary periodically to restore their time relationships in order to keep their divergence within acceptable limits.

5. Interconnection options

In a layered system, such as that depicted by the transecting layers of the system model in *Fig. 4*, it is possible to use any of several interconnection solutions at each layer of the stack. The total number of possible implementations of any portion of the system for which individual selections can be made then becomes the product of the number of choices at each layer. When there are more than a couple of choices for each layer, the number of potential system design combinations can quickly become staggering.

Understanding the confusion this could bring, the Task Force developed the concept of a matrix of preferred implementations. The matrix listed an array of applications for which implementations would be required and then indicated one or two combinations of choices at each layer that were to be preferred over the other possibilities. It was not possible to complete this work within the time-frame of the Task Force's efforts. Hence, only a general outline was provided, along with a number of definitions, in the expectation that the continuing work within the SMPTE would complete the task. The goal is to provide the industry with a series of templates that can be used for the different applications that will arise.

Some of the definitions of transfer types that came out of the work on implementations can help us to differentiate between the types of applications that may require different solutions. In particular, the concepts of "hard real-time," "soft real-time," and "non real-time" should be understood. Real-time here means that the transfer occurs in the actual time in which a process occurs, i.e. at 1 x play speed.

Hard real-time operations must happen *at* a certain time; there is no possibility to repeat and there may not be a chance to re-do (as, for instance, in the case of a live event). Hard real-time operations demand the highest priority.

Soft real-time operations must happen *by* a certain time, and there may be an opportunity to re-do. When the time allotted for soft real-time operations runs out, they become hard real-time.

Non real-time operations need not be completed within any time constraints. They are typically file transfers that can be either faster or slower than real-time.

6. Migration

The Systems subgroup recognized that making the proposed change in the approach to control mechanisms is revolutionary in concept. It requires the adoption and widespread use of standards in an area that previously has seen little use of the standards that existed, however painstakingly they were developed. Thus the subgroup conceived a migration strategy that would serve (i) to make the transition reasonable to accomplish and (ii) to build confidence among manufacturers – in particular that their investment in using standards for control would reap benefits in the long term. This was coupled with a recognition that the manufacturers of control systems are starting to realize that they derive no

benefit from having to write drivers for each new device that comes on the market; their added value comes instead from the application and the user interface, which is where they should concentrate their efforts. The increasing complexity both of future systems and of future controlled devices will make a standardized set of protocols increasingly attractive for all.

The first step along the migration path is the placing of all existing control protocols in the public domain and making them accessible through a single point of contact. Using the Internet for accessibility will permit developers to obtain accurate information on protocols and will also provide a mechanism for disseminating information about protocols, even after their developers have discontinued supporting them. This step will build confidence in the concept of open control standards and will demonstrate an industry commitment to the process.

With existing protocols in the public domain, it will be possible to develop a set of *Essential Common Protocols* that build on the existing protocols, and an *Object Reference Model*. The Essential Common Protocols will, in turn, serve as the basis for a *Distributed Object Model* mechanism for control. Once these pieces are in place, it will be possible to interconnect both new equipment that makes use of the object-model approach directly, and older equipment that does not. This can be done through the use of control systems that support both the old and the new protocols, controlled devices that support both the old and the new protocols, or proxies that translate protocols.

7. Economic model

The history of the broadcasting industry has been one where the costs of both development and support of equipment are built into the initial sales price of the hardware, while the software – including maintenance and upgrades – is provided at low or no cost. When manufacturers wanted to update features, they sold completely new units, even when in reality only the software was being changed. With the value and cost of digital equipment now primarily lodged in the software rather than the hardware, other industries (such as the IT industry) have long since used a different model – in which the hardware and software are sold and supported separately.

The Task Force perceived that the current economic model in the broadcasting domain no longer serves the industry well. It causes hardware to be replaced when it is not necessary to do so. It fails to recognize that the value in products is now dominated by the software. It also fails to recognize the real costs of developing and supporting the software. All of this results in much more expensive equipment, which must be capitalized at the beginning of its life cycle when the cost of money generally is highest, thereby compounding the discrepancy.

The Task Force therefore proposes the adoption of the computer industry model in which hardware and software are unbundled. This is expected to encourage better software support from the manufacturers, by rewarding them for upgrades and maintenance. It should be attractive to users because it will lower the initial cost of the equipment and spread the total costs of ownership over the life of the equipment, thus reducing the overall financial impact.

8. Standards

As in the rest of the work of the Task Force, the ultimate focus of the Systems subgroup was on the eventual development of standards. It is expected that the standards development effort will largely be carried out by the SMPTE. To that end, a series of requirements for standards at the systems level is included in the Final Report. It is divided into two categories: those items that were sufficiently well understood that they could be turned over to the SMPTE at the earliest possible time for standardization work to commence, and those items that required more development within the Task Force before being turned over to the SMPTE. These latter items have now been given to the SMPTE with the release of the Final Report, and the SMPTE is in the process of taking them over as standardization projects.

To help maintain the close working relationship that has developed between the EBU and the SMPTE during the Task Force effort, the SMPTE has also brought several highly-regarded EBU participants into leadership roles within the SMPTE. This will allow the EBU to help in guiding the work going forward and will assure continuing close liaison between the two organizations.

Compression

H. Schachlbauer
IRT

The Compression subgroup of the Task Force was set up to provide guidance for the longterm integration of compression into programme production. Its work led to the conclusion that there is no single member of a compression family that satisfies the requirements of a fully-networked digital production facility.

The two compression families ultimately selected by the subgroup were MPEG and DV, each of which offers individual trade-offs in terms of coding flexibility, product implementation and system complexity. This article gives an overview of the factors which led to this choice.

1. Introduction

Compression is a key enabling technology in the context of a fully-networked digital environment – particularly from an economic viewpoint. Through the use of compression, broadcasters hope to achieve significant cost savings in the areas of data transfers, storage, etc. No broadcaster, after all, wants to compress just because of the beauty of compression!

There is a very complex relationship between the quality we want to maintain in our broadcasting assets and making that quality truly predictable after a lengthy post-production process. The ultimate quality obtained from a digital production system is dependent on:

- the data-rate;
- the complexity built into the compression system;
- the type of network and the bandwidth available to transport the datastream between the various storage devices and network devices.

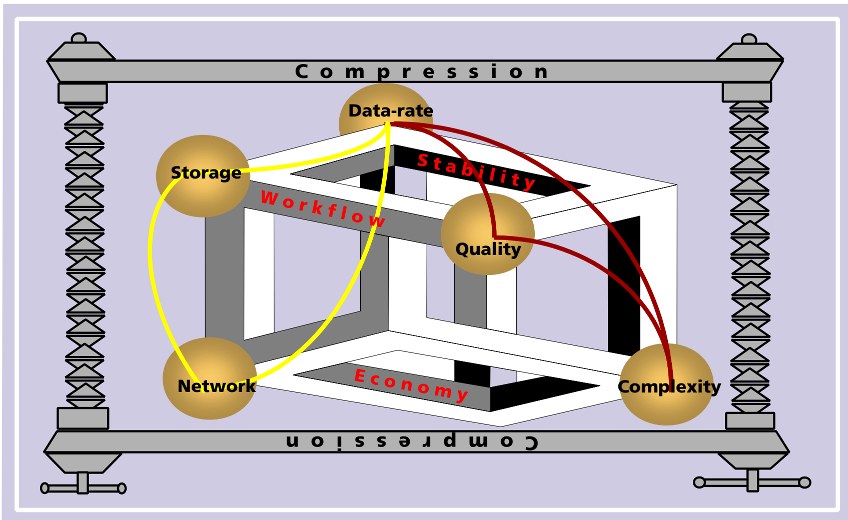

Figure 1
Some of the important issues associated with compression.

The major objective now, in terms of a compression system, is to find an optimum balance between an achievable signal quality which will hold for the next ten to fifteen years, and an economic model which will make networking and storage a viable proposition.

2. Compression considerations

The Compression subgroup was confronted with a range of different and, of course, incompatible compression systems which it had to evaluate and appraise on a piece-by-piece basis.

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The following criteria were used:

- the ultimate technical quality that could be achieved, versus the data-rate required for that;
- interoperability between compression schemes using, for example, different coding parameters;
- the editing granularity versus the complexity of the network editing control.

We had to verify how all the availalable compression schemes complied with the requirements we set out in the first Task Force report (April 1997) – namely, and most importantly, the format stability. We did not want to consider formats which would not survive the day, or the next ten years. This meant that the chip-sets had to be provided at an economical price and on an equitable basis. They would have to be standardized, and all the elements required to reproduce the compression system and all the modules pertaining to it (e.g. the mapping into various networks) would have to be laid open, so that any manufacturer could build his equipment from these chip-sets.

Abbreviations

4:2:2P@ML (MPEG-2) 4:2:2 Profile at Main Level (Professional MPEG)

GoP Group of pictures

IEC International Electrotechnical Commission

ISO International Organization for Standardization

ITU-R International Telecommunication Union, Radiocommunication Sector

JPEG (ISO/IEC) Joint Photographic Experts Group

M-JPEG (ISO) Motion - Joint Photographic Experts Group

MP@ML (MPEG-2) Main Profile at Main Level

MPEG (ISO/IEC) Moving Picture Experts Group

NLE Non-linear editing

SDI Serial digital interface

SDTI Serial data transport interface

SMPTE (US) Society of Motion Picture and Television Engineers

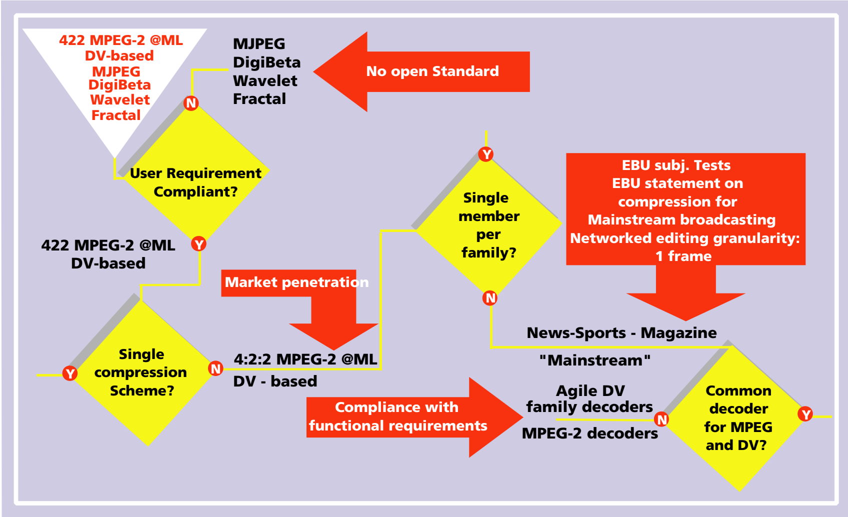


Figure 2
Flow chart used to select suitable compression schemes.

We had to look at the picture-quality ceiling available with the different compression systems proposed for today and tomorrow. We also had to look at the availability of integrated decoders and intra- and inter-family agile decoders (which are explained later), and of course we had to look at the pros and cons of choosing a single compression system rather than a whole compression family.

We also looked at the format development potential because, obviously, we did not want to promote a compression scheme having a lifetime not exceeding five, six or seven years – we wanted one which would form a solid basis upon which systems could be built in a very compatible way.

We had to identify possible problems in the area of interoperability and complexity, and we did focus on near-term solutions (a very essential element) because we were very well aware of the fact that broadcasters would implement these systems on a piecemeal basis.

It was quite clear from the beginning that we would have to provide a migration path for broadcasters to proceed from where they are now (i.e. using analogue systems and partially-digitized plant) towards the fully-networked digital environment. At the same time, broadcasters would wish to be able to add elements as and when they wanted to, safe in the knowledge that they could build one element on top of another without making the previous element obsolete.

And of course the subgroup looked at proposed solutions which required a whole network to be in place to make them really work. While we are all hoping that the network approach will actually happen, we are not promoting solutions which are based purely on that approach.

3. The decision process

The subgroup studied various compression schemes, their options and quality ceilings. As we wanted to base our decisions on experimental evidence, not just on gut feeling and a crystal ball, there had to be underlying evidence as to how these proposed schemes would be usable within a broadcast environment (we did request formal written commitments for standardization in a number of areas). In addition to this, given that VTR and disk will continue to coexist for a long while yet, only compression schemes which would also address the problem of recording on a VTR were considered.

The subgroup was eventually confronted with six different compression schemes, ranging from professional MPEG, to DV-based, Motion-JPEG, Digital Betacam, wavelet encoding and fractal encoding. Applying our user requirements, we had to eliminate a number of these schemes because they simply did not comply (see *Fig. 2)*.

Motion JPEG was not standardized, and consisted of many variants – although recently an effort has begun within the SMPTE to have M-JPEG made compliant within itself and inter-operable with M-JPEG equipment from other manufacturers, but that enterprise is just starting.

We approached the proponents of Digital Betacam, which has a very successful compression format hidden away within the equipment. However, they said that they would not open the compression system to the public domain for standardization, and that they had no intention of using this compression format for networked operation. That rendered Digital Betacam a non-contender.

Wavelet compression is something which works, but is not yet standardized.

Fractal compression is successfully used in environments where the imbalance between encoding and decoding complexity can be tolerated – in graphics, for instance. However, it is not something that could be universally used in a post-production environment.

So what we were left with were:

- MPEG-2 4:2:2 P@ML (abbreviated to "Professional MPEG" in the accompanying figures), which is an adaptation of the MPEG-2 MP@ML;
- a range of DV-based compression schemes.

At the time, there were only two commercial systems on the market we could base our decisions on – one was Betacam SX and the other was DVCPRO.

So the question then arose - *is it going to be just one of those two systems, or is it going to be both?*

It was quite clear from information about market penetration that this idea of a single system covering the whole range of different applications within post-production was a non-starter. So we had to recognize that we would be facing two different compression families in the future, one based on MPEG and the other based on DV.

This lead us to another important question – *will these two systems, SX and DVCPRO, cover the whole range of quality expectations we normally have in television production?*

4. EBU subjective tests

The EBU has carried out a range of tests to try to answer that question. It has found that for applications such as news, sports and magazine programming, these two formats deliver *almost* equal quality. Hence, for news, sports and magazine programmes, we can happily live with MPEG- or DV-based compression in the range 18 - 25 Mbit/s.

This could have led us to the conclusion that we would need just one member of either compression family, but this was not to be. We identified a second quality area – the whole area of mainstream television – which demands compression at around 50 Mbit/s.

Wouldn't it have been nice to have a *common decoder* for these schemes! But the manufacturers at that time said "no, it's not going to happen" so we have ended up with two families of compression and two different *agile decoders*, both of them able to handle the compression members of their own family only.

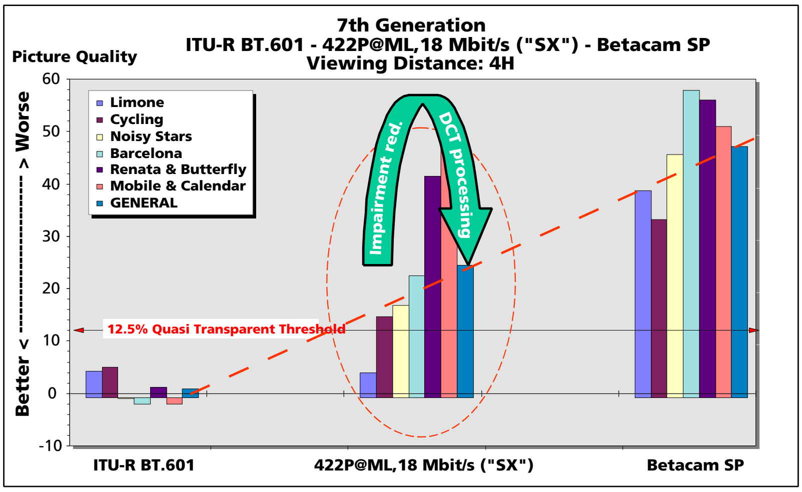


Figure 3
7th generation picture quality at a viewing distance of 4H: 4:2:2P@ML, 18 Mbit/s ("SX").

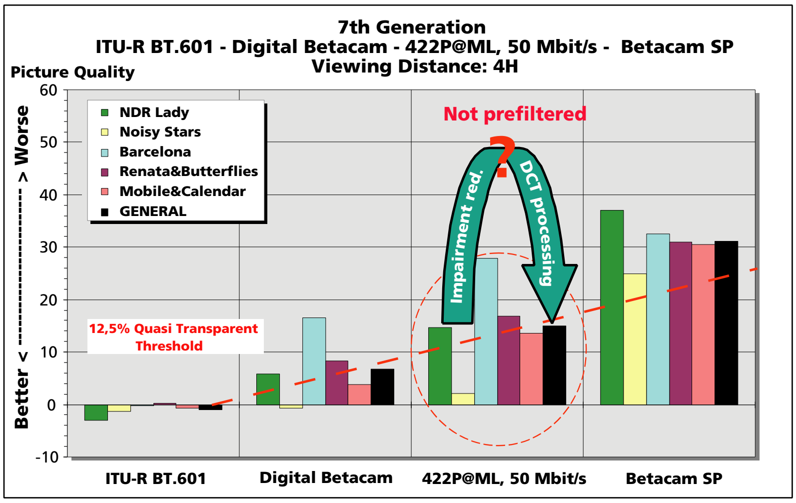
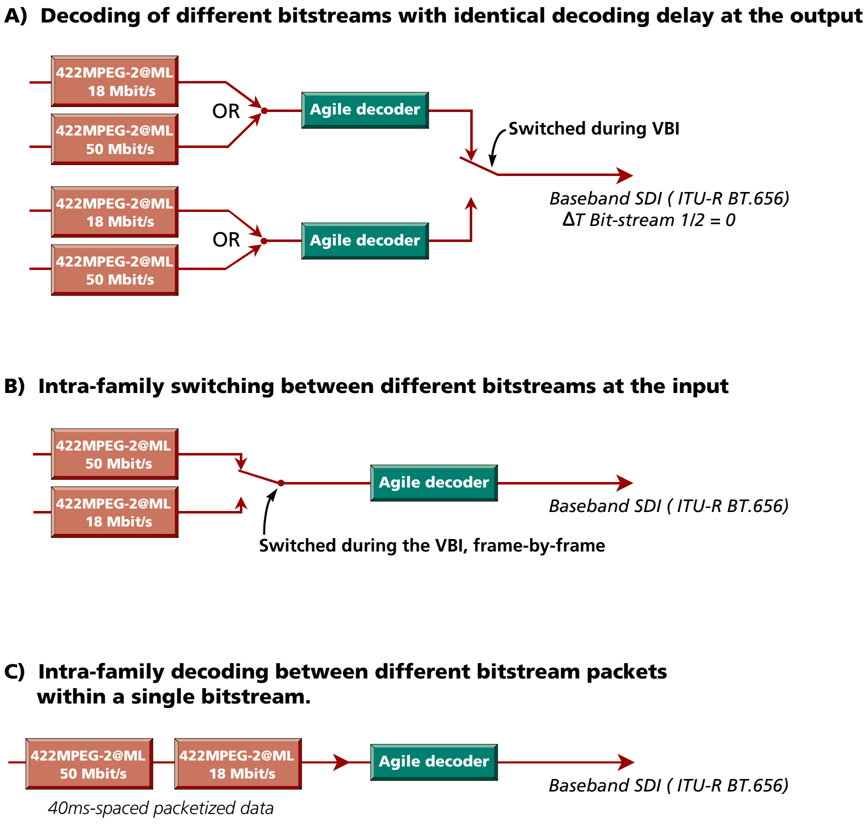


Figure 4
7th generation picture quality at a viewing distance of 4H: 4:2:2P@ML, 50 Mbit/s ("Professional MPEG").

NOTE: The different behaviour of Betacam SP that is apparent in *Figs. 3* and *4* is due to the fact that two different versions of the SP equipment – in different states of (mis)alignment – were used for these tests. It thus indicates, to some extent, the practical quality range of the Betacam SP format.



Native Decoders

Native decoders which have been designed to operate on non-standard bitstreams, e.g. for optimized stunt-mode performance (shuttle, slow-motion) or for special functions, are acceptable. The decoder chip-set should be available on a non-discriminatory basis on fair and equitable conditions. Details of possible deviations from the standardized input datastream should be in the public domain.

Figure 5
Some applications of agile decoders within a single compression family

Fig. 3 shows the subjective performance of the SX system when tested at 18 Mbit/s in accordance with the well-known ITU-R Recommendation BT.500. It clearly shows that, at the seventh generation under worst-case conditions, we end up with conspicuous artefacts. However, studies are still going on to find how we can mitigate these aberrations by aligning macroblocks throughout the post-production process, by means of auxiliary information carried within the bitstream. Perhaps, this will provide a solution in the future – for both MPEG and DV compression.

Fig. 4 shows the subjective performance of what we call the mainstream television profile, running at 50 Mbit/s. As you can see, the average quality obtained at the seventh generation, under worst-case conditions, is just slightly above the level of visibility. Once again, in the future we may be able to reduce these artefacts by aligning macroblocks throughout the post-production process but, until then, our final television products will be maimed with the artefacts caused by multiple coding and decoding.

One other option that both families can exploit – at least those compression formats which use temporal redundancy – is the fact that you can alter the GoP structure along the post-production process, in an attempt to maintain some of the original coding information. For example, you can change to a different GoP structure in order to reduce the datarate for increased storage efficiency and then revert to the original GoP when loading this content from the storage device into a different device. This technique is currently being investigated in various places – there is a degree of enthusiasm about it – but we still have to await the results before making a positive public statement about it.

The subgroup's deliberations were very conservative because we had to establish a base-line commonality. It is a worst-case scenario – which can only be improved upon. As technology evolves, some of the differences that we noticed between the two systems will possibly disappear, or will at least be smaller than we originally thought they would be.

5. Agile decoders

An *agile decoders* is any decoder within a particular family, be it MPEG or DV, that could cope with a range of compressed input signals (see the examples shown in *Fig. 5*). In some cases it would be necessary to use interframe switching between the different bitstreams at the input, in which case we would then end up with different data packets within a single SDTI bitstream. The agile decoder would have to be able to output that bitstream for further multiplexing, without causing any hiccups.

It must be acknowledged that agile decoders will only be able to work with fairly standard bitstreams. They will not, for example, be able to cover things like faster than real-time play-out, pictures in shuttle or stunt modes and other similar applications which require a different arrangement of the packets in order to optimize the visualized result. These specialist applications will require *native decoders*, designed by the equipment manufacturer for optimum decoding quality.

Fig. 6 gives a visual representation of the problems encountered through the use of two different compression families. There are three distinct planes on the diagram:

- a DV plane which represents the whole DV family, and this is not just one company and one tape – there are at least three tape formats currently in use;
- an MPEG plane which covers the range of available GoP structures;
- a processing layer.

Ideally the two compression layers should interoperate at an SDTI compressed bit-rate level. However, that is not happening just yet and it is necessary to transmigrate from the SDTI layer into the processing layer which is still SDI. Taking into account the artefacts that are normally caused by coding from A to B and from B to A, what we are trying to achieve as time goes by is to carry out a number of these processes within the compressed area itself. Thus, if there is an economic implementation that will allow us to do this, then that is certainly the way to move forward. At the moment we are still handicapped by having to go from the compressed to the uncompressed state in quite a high number of cases, leaving us with a very conspicuously impaired result at the end.

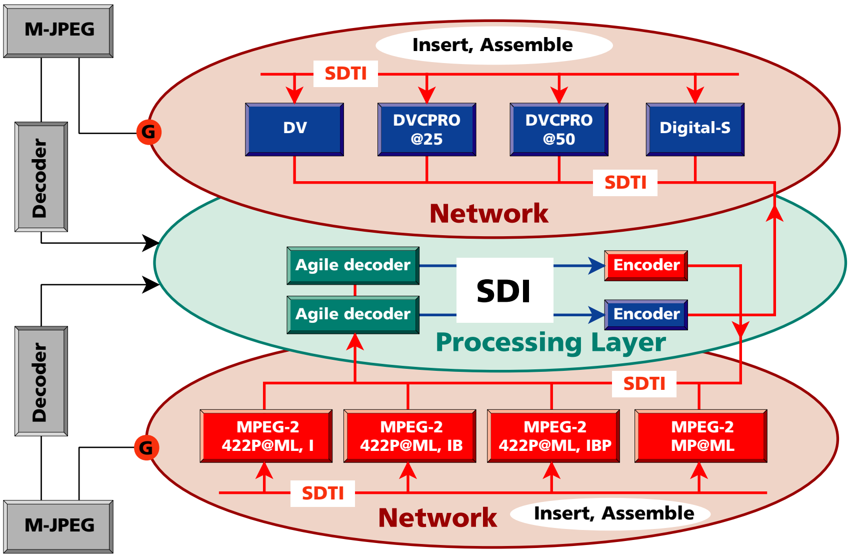


Figure 6
Model showing the two chosen compression families and the need for a processing layer.

There are gateways for M-JPEG which we have not promoted as having a future in a fully-networked environment. This has astounded and surprised some people because it is obvious that Motion-JPEG is everywhere in applications based on hard disks, and NLE in particular. It is now up to the proponents of M-JPEG to provide the appropriate gateways into the networked environment – either into DVCPRO or into MPEG. The way into SDI is open for everybody, that is quite clear.

As time goes by, the differences between the DV and MPEG formats will hopefully disappear. Devices will be provided to enable the agile decoding of both formats, thus allowing a controlled mixture of both formats whilst giving a predictable output signal at the end of the production process.

__Horst Schachlbauer__ graduated in Telecommunications from the University of Munich in 1967 and has since worked for the IRT, the central Research Laboratory of German, Austrian and Swiss public broadcasters.

*Mr Schachlbauer been very involved in the development of standards for digital television on national as well as international platforms, e.g. ITG, EBU, CCIR and ETSI. In particular he was involved in the specification of CCIR Rec. 601, the D-1 recording format, the **S**erial **D**igital **I**nterface and PALplus.*

Currently, Horst Schachlbauer heads the EBU MAGNUM committee which closely liaises with manufacturers in the area of recording technology for television. He also chairs a number of national and international Project Groups dealing with digital television production technology and archiving. He served as the European co-chairman of the EBU/SMPTE Joint Task Force and has recently been elected a Fellow of the SMPTE.

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Wrappers and Metadata

O. Morgan
Avid Technology

The Wrappers and Metadata subgroup of the Task Force set out to find a single comprehensive solution which would cover the requirements for classifying Metadata, and the requirements for wrapping programme Content into suitable containers which would ensure complete interoperability in a future networked production environment.

As described in brief here, this work has led among other things to the creation of a Metadata "encyclopaedia", which is maintained by a registry mechanism, the specification of a Unique Material Identifier for the Content contained in a Wrapper, as well as the specification of various Wrapper formats for the streaming and storage of Content.

1. Metadata

Metadata is broadly defined as "data about data" and the number of distinct varieties of Metadata is potentially limitless. It can, for example, be process-oriented, business-oriented or data-format-oriented.

The SMPTE is working on the definition of standardized packet formats to encapsule Metadata such that it can be stored within and transferred throughout digital television systems, without the need for it to be translated or re-keyed every time it is interconnected. *Fig. 1* shows some of the proposed packet formats that are being worked upon and brought together into a single unified architecture. There are individual items of Metadata, groups of Metadata and packs of Metadata – which are familiar concepts in the world of data-processing.

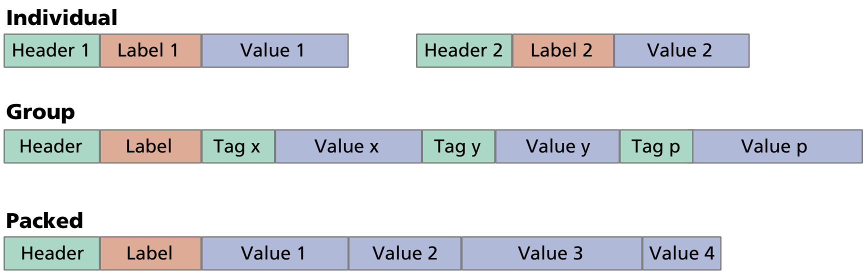

Figure 1
Some examples of the proposed packet formats for Metadata.

Each item of Metadata incorporates an SMPTE *Universal Label* which, at this point in time, is in a standardized ANSI format. These labels are all entered into an SMPTE registry which can be thought of as a "table of contents" to some huge "encyclopaedia". This encyclopaedia contains, for example, all the relevant specifications in the world, all the standards that relate to our industry, and all the things that describe and tell you what is the meaning of a particular piece of Metadata, or a piece of Data Essence, or a piece of Video / Audio Essence.

The SMPTE registry has a couple of interesting features: one is that you can keep adding new labels to it, and the other is that it does not necessarily provide a full description of the items which it is linked to in the encyclopaedia – it keeps these items all separate so that, for example, you can find out where to look in order to pull out one or more required specifications and use them. The registry also enables the stored specifications to be placed within the public domain so that, for example, if "manufacturer A" makes a specification public by means of the registry, then "manufacturer B" or "system C" can openly use that specification.

The SMPTE has established the SMPTE *Registration Authority* which is an electronic global library, index and table of contents for all the relevant specifications which now exist and for the ones which may be added in the future. It uses ISO standard rules for the operation of registration authorities, providing a global dictionary of Metadata items. Because it is extensible, it allows

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a “fast-track” approach to extending the existing standards. This is something which is very important because the standards process has had a reputation for moving much too slowly for the rapid advances that take place in technology.

2. Wrappers

When you combine together all these bits of Metadata with all the bits of Essence (data, audio and video), the result is *Content*. This Content must then be put inside *Wrappers* (i.e. containers or file formats) so that it can readily be moved and stored throughout the system.

It swiftly became clear to the Task Force that there is not just one magic container format. There are a lot of them, each one optimized for a different purpose – perhaps optimized for streaming, maybe optimized for storage, or perhaps optimized for databases and so on. However, there is a generic data model which can be put into any of those containers.

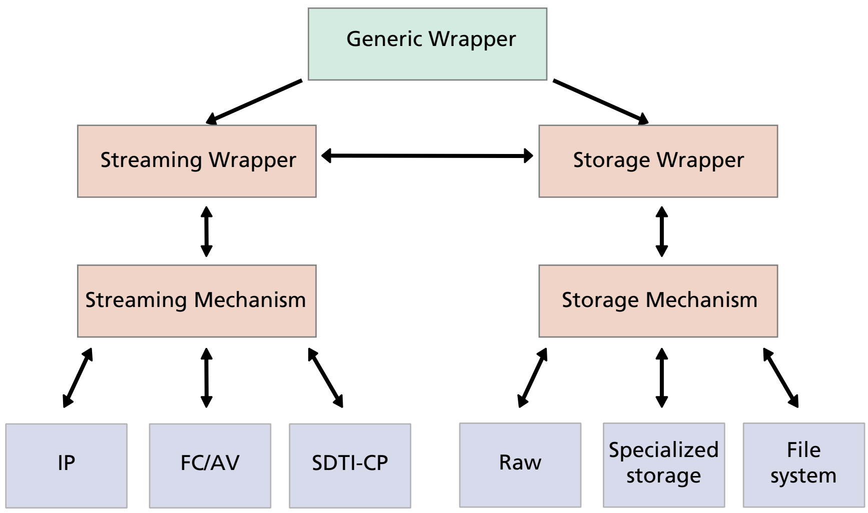

Figure 2
Example of a generic Wrapper format.

The type of things you put into those containers are: Metadata, Essence formats, relationships between bits of Metadata, and the relationships between Metadata and Essence. Consequently, you end up with many physical representations (many container formats). But the outcome is the all-important interoperability made possible by the Metadata and the Registry, as well as by the templates which determine how items are used in specific applications.

2.1. *Generic Wrapper formats*

Fig. 2 gives an overview of generic Wrapper formats, with specific varieties targeted towards *streaming* purposes or *storage* purposes by means of streaming mechanisms (interconnects) or storage mechanisms. The diagram shows specific examples of target applications:

- IP-based;
- Fibre Channel AV-based;
- SDTI-based;
- raw content, such as recorded on tape;
- specialized storage systems for advanced access methods in computer systems;
- normal file systems such as can be found currently in server systems and computer systems.

This list is by no means complete. Standardization projects aimed at container formats are under way both within the SMPTE and elsewhere, e.g. within NCITS, ANSI, X3, ISO, MPEG and

Abbreviations

AES	Audio Engineering Society
ANSI	Americal National Standards Institute
AV	Audio-video
DAVIC	Digital Audio-Visual Council
FC	Fibre Channel
IEC	International Electrotechnical Commission
IP	Internet protocol
ISAN	International standard audio-visual number
ISO	International Organization for Standardization
MPEG	(ISO/IEC) Moving Picture Experts Group
NCITS	(US) National Committee for Information Technology Standards
SDTI	Serial data transport interface
SMPTE	(US) Society of Motion Picture and Television Engineers
UMID	Unique material identifier
UPID	Unique programme identifier

Oliver Morgan *studied Physics at Oxford University. In 1978, he entered the TV industry in the UK and ever since has worked primarily on software development for post-production. He was Director of Engineering at Convergence Systems until 1990 and Manager of Technology at the Sony Advanced Development Center in San José, California, until 1997. Since then, he has been a Senior Consulting Engineer at Avid Technology in Massachusetts.*

Mr Morgan, a holder of eight patents in TV production and related technologies, was previously Chairman of the SMPTE Working Group on Editing Procedures, and Chairman of the SMPTE Committee on Wrappers and Metadata Technology. He has been a regular contributor to SMPTE conferences and is a Fellow of the SMPTE.

More recently, Oliver Morgan chaired the Wrappers & Metadata subgroup of the EBU/SMPTE Task Force.

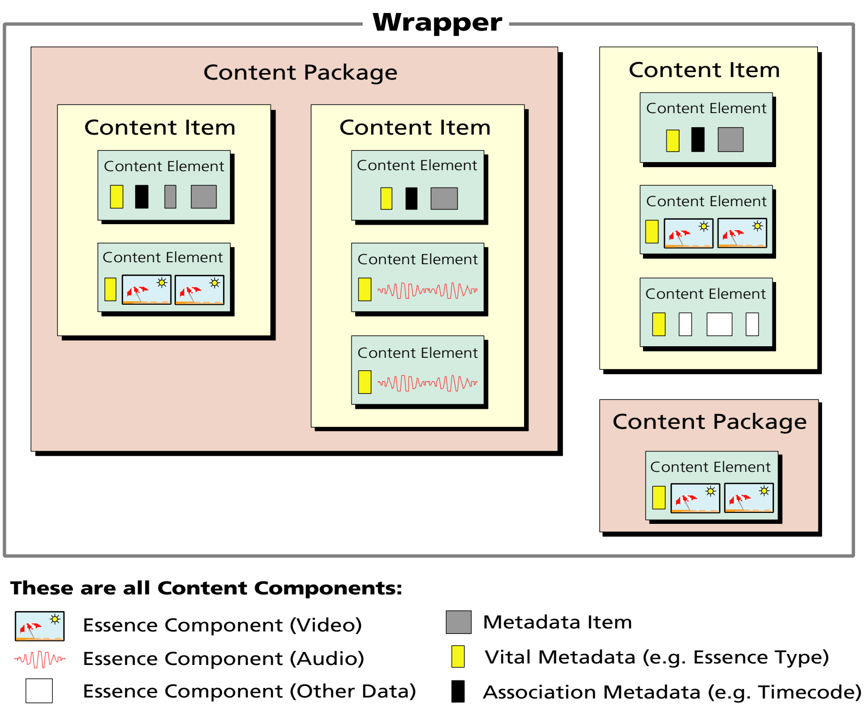

Figure 3
Example of Simple Content Packages, Items and Elements within a Wrapper.

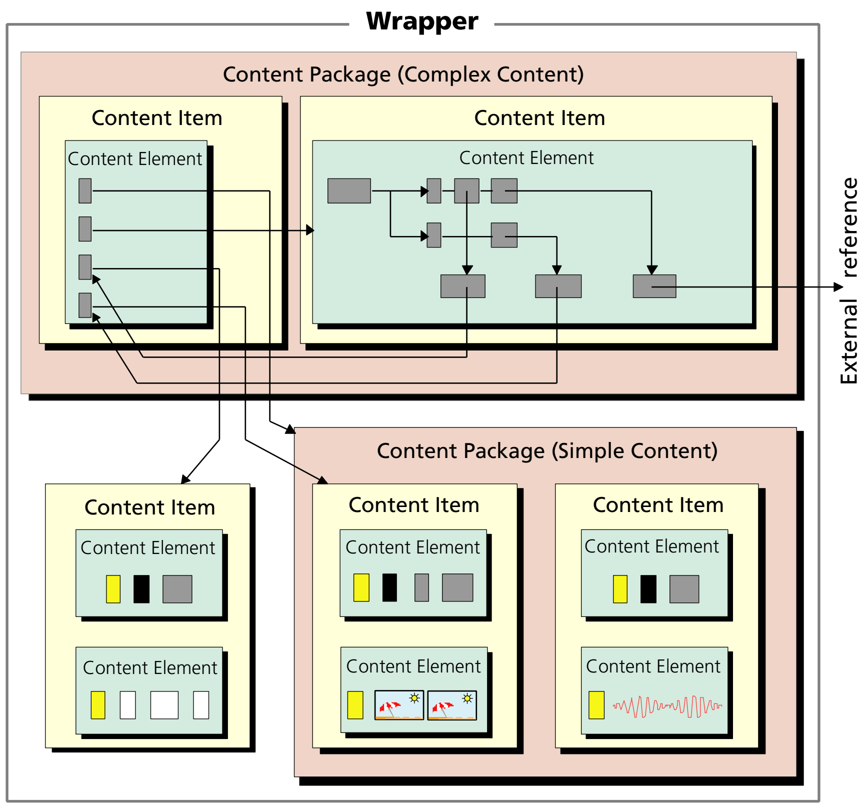

Figure 4
Example of a Complex Content Package.

the AES. What we are attempting to achieve through the work of the Task Force is to have all of these formats basically able to contain elements of the common data model.

2.2. *Simple content packages*

Fig.3 shows the simple content model that has been derived by the Task Force, starting from the work of DAVIC. What we have here is content which is organized into *content packages* which are made up of *content items* which, in turn, are made up of *content elements*. There are different types of content elements including Video Essence, Audio Essence, Data Essence, Vital Metadata and Association Metadata.

Familiar items of Content – such as video, audio, captioning, virtual studios, scripts and timecode – can be described very easily by *Template* mechanisms which are available publicly via the SMPTE registry. Thus the right template for a specific application can readily be found.

These simple content packages represent a very good way of organizing programme contributions and delivery.

2.3. *Complex content packages*

Complex content packages are somewhat different to the simple packages described above. In this case, what we are trying to put into the container is all the data, all the choices, all the possible revisions, all the undo history and all the 3D models that go to make up elements of a programme. So we end up needing a very comprehensive index of everything we have inside the package; the content itself and the composition Metadata. We then find that we have got all kinds of hierarchical and recursive rules about what we can put inside these packages.

Fig. 4 shows a basic example of a complex content package which consists of some simple content, some composition Metadata (which looks a little bit like an edit list) and an Index. Such a package may contain edit decision lists, news items, news stories and even complete shows. However, while this kind of content package would be good for production, content creation

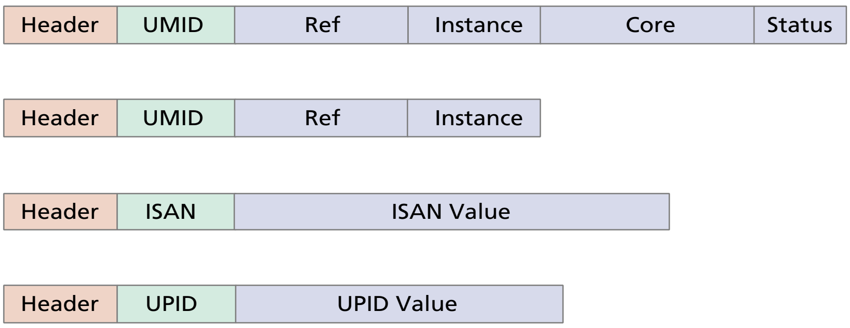

Figure 5
Examples of Unique Content Identifiers.

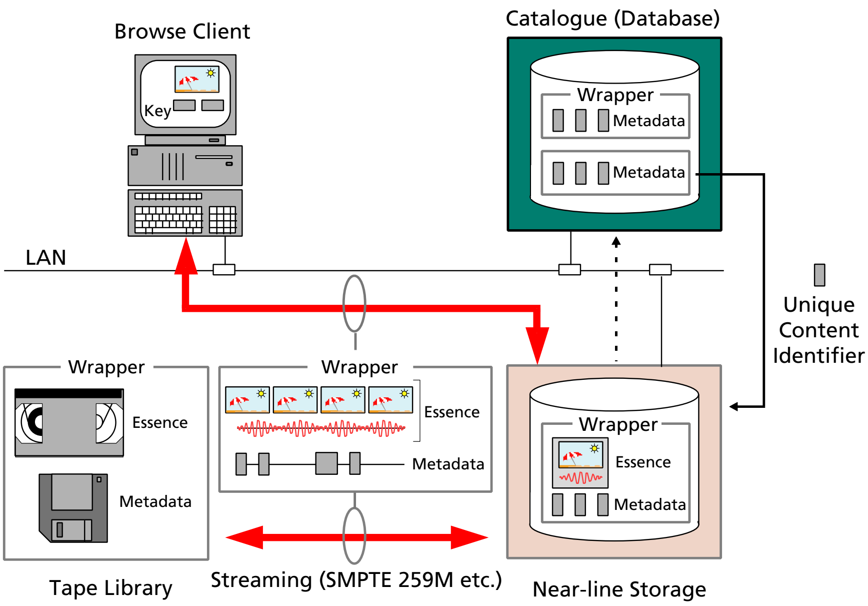

Figure 6
Example of a future network-based production facility.

and editing, it is not really intended for delivery.

2.4. Unique content identifiers

Unique content identifiers basically come in two types. One is an identifier for finished programmes, the other is an identifier for the content objects which go to make up those programmes. The essential point about a unique content identifier is that it identifies the piece of content independent of its location. This allows, for example, multiple copies of the content to be referred to by an asset management system, and it also enables hierarchical storage, content browsing, online tape libraries, archives and many other such applications.

There are different types of content identifiers for different uses. *Fig. 5* shows some of the diverse examples (e.g. Unique Material Identifiers, Unique Programme Identifiers) which are being brought together into a single standard framework for unique content identifiers. Because a unique content identifier starts off with the SMPTE Universal Label, which is a registered entity, each identifier can be used by a system whose natural content identifier is different – the Task Force's unique content identifier is an interoperable, interchangeable entity.

Fig. 6 shows some of the areas that these standards and specifications touch upon, even if the diagram presents a rather simplified view of a networked studio of the future. At the bottom left of the diagram, content is entering the system by the traditional method, i.e. via a video tape and a data storage device. It has a wrapper around it – we can think of this wrapper as a rubber band which joins together the video cassette and the floppy disk (or a piece of paper) which contains the Metadata. The incoming material is carried via an SMPTE standardized streaming wrapper into the near-line storage area which uses file-system storage wrappers containing the Essence and the Metadata (i.e. the Content).

The upper right part of the diagram shows the catalogue, or database, where some of the incoming Metadata is stored and organized in a database format, for ready access and use by the exploitation part of the system. The separate database and server aspects of this area are tied together using either unique material identifiers or unique content identifiers.

And finally, in the upper left part of the diagram is the browsing client. It uses other forms of streaming wrappers and other forms of APIs to access the material held within the system.

3. Standardization

In conclusion, there is at least one SMPTE project to write a standard for each aspect and for each element of the Wrappers and Metadata issues that were studied by the Task Force. We are also aware of similar projects in other standards bodies, and we are closely liaising with these bodies in order to keep everyone within the same interoperability framework.

Networks and Transfer Protocols

H. Hoffmann
IRT

The Networks and Transfer Protocols subgroup of the Task Force had the responsibility of finding the best technologies to enable different data types to be moved around a networked production environment. It had the task of identifying the best methods for (i) audio/video streaming in real-time (and faster than real-time), (ii) file transfer (also at different speeds) and (iii) file access.

The chosen methods should guarantee the interoperable transfer of programme content between devices, and these transfers should meet the high-end requirements of the TV broadcast world. An additional part of the subgroup's work was to identify and define the further work that needs to be carried out by standardization organizations.

1. Introduction

The Networks and Transfer Protocols subgroup issued a so-called Request For Technology (RFT) to the different parts of the industry concerned with networks, interfaces and transfer protocols which can be applied to our television environment. We received a variety of responses:

- *Advanced Streaming Format* (ASF), from Microsoft Inc.;
- *QuickTime*, from Apple Computers Inc.;
- *Fibre Channel Audio/Video*, from the Fibre Channel community;
- the *ATM Forum* sent a representative to participate in our meetings;
- the proponents of *SDTI* became heavily involved in our work.

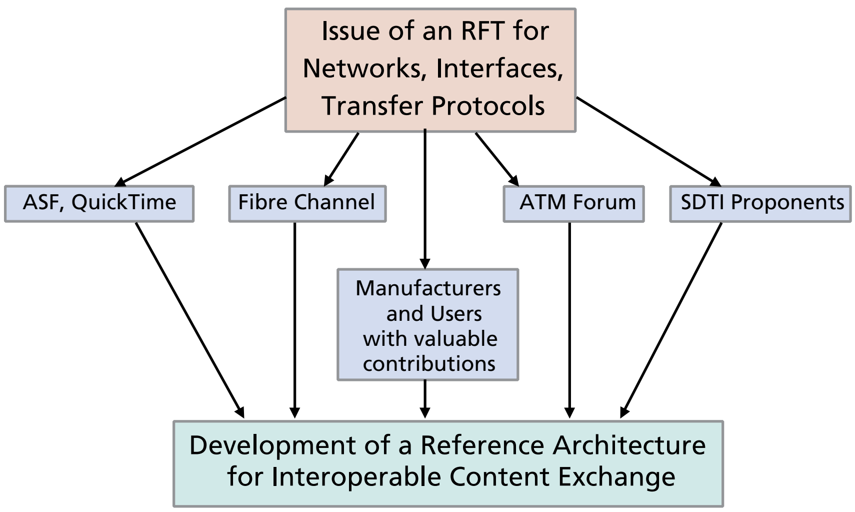

Figure 1
Workflow of the subgroup.

The manufacturers and users, all of whom made valuable contributions to the work, finally came to the definition of a so-called *Reference Architecture* (RA) as shown in *Fig. 1*. Al Kovalick of HP should be given special mention here because he had the idea of the RA for interoperable content exchange. For the RA, we had to select a few suitable technologies from a universe of offers submitted by the computer and related industries. For the streaming of content in real-time and faster, we had to consider the mapping of different applications into the available transport mechanisms – for example, DV into ATM, MPEG into Fibre Channel. We also had to recommend a limited number of interfaces from the growing list of candidates: Ethernet, Fast Ether-

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net, Gigabit Ethernet, FC, SDTI, SDI, ATM, different flavours of ATM, etc.

We had to identify a number of interfaces and transfer protocols for general-purpose use and low-performance applications, and a number of interfaces and transfer protocols for very high-performance use, specialized applications and, of course, for coping both with remote and local transfers.

2. Streaming

Streaming means that you have to "push" content across channels and networks in a point-to-point or in a point-to-multipoint topology. This is similar to the way in which we currently handle our broadcast content.

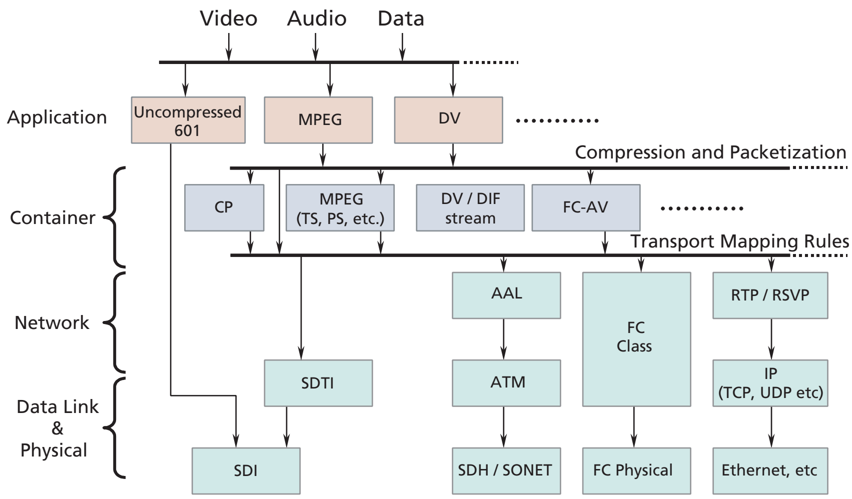

Figure 2
Streaming technologies.

The important requirement here is to transfer the content in real-time, which is not easily met by interfaces such as FC or ATM.

The links for streaming are often uni-directional, and the receiver should be able to "join" a stream which has already started. The consequence of having uni-directional links is the bounded quality of the received signal – there is no mechanism in the receiver to flag a corrupted signal, which means that we have to apply certain Quality-of-Service (QoS) parameters to the link in use. Amongst these parameters are the bit-rate, the jitter and wander, the transmission delay and also synchronization issues.

So which technologies have we identified? In the physical domain (see *Fig. 2*), SDI is of course very important for internal studio applications. SDH and SONET are very important for wide-area transmissions; so too are the physical layer of Fibre Channel, and of course Ethernet, Fast Ethernet and Gigabit Ethernet.

In the Data Link and Network area, SDTI is one of the hot topics concerning the transmission of compressed signals within the studio. ATM is very suitable for WANs, and we should not ignore FC and IP.

What was urgently required was a certain type of mapping rule to enable us to map programme containers such as DV, MPEG and FC A/V into the transport mechanisms mentioned above. We have identified a couple of mappings whose standardization has already been completed. However, other identified mappings will have to be further worked on, in the follow-up activities of the SMPTE.

The subgroup has generated the following recommendations within the RA for streaming (and this is not an exhaustive listing):

- **SDTI** is currently the choice for internal studio interconnects. This means that in order to transmit compressed signals without re-encoding within the studio, SDTI is the right choice – if you have to do it in real-time and at faster than real-time.
- **Fibre Channel** is a high-performance file transfer mechanism. The FC A/V project is also working on a streaming implementation, but this is only on paper for the moment. So we encourage and require all the supporters of FC to implement this standard for streaming in the form of real hardware products.
- **ATM** is the choice for WAN streaming – no doubt about that. However we have identified that there is still a problem, especially between North America and Europe. The AAL1 and AAL5 issue needs to be sorted out. We also need a guideline on how the wander and jitter issues can (probably) be solved.

Gateways are needed between the studio and the WAN - for example, how can we move SDTI payloads over wide-area networks?

The Final Report of the Task Force includes a mapping table (in Section 5.7.3.) which was developed by our subgroup. This table shows (by means

Abbreviations

ANSI	American National Standards Institute
ASF	(Microsoft) Advanced Streaming Format
ATM	Asynchronous transfer mode
A/V	Audio / video (visual)
FTP	File transfer protocol
IEC	International Electro-technical Commission
IEEE	(US) Institute of Electrical and Electronics Engineers
IP	Internet protocol
ISO	International Organization for Standardization
FC	Fibre Channel
MPEG	(ISO/IEC) Moving Picture Experts Group
QoS	Quality of service
RA	Reference architecture
RFT	Request for technology
SDI	Serial digital interface
SDTI	Serial data transport interface
SMPTE	(US) Society of Motion Picture and Television Engineers
TCP/IP	Transmission control protocol / Internet protocol
WAN	Wide-area network

of a "C" for "complete") which of the transports are already defined. For example, we have an MPEG transport stream mapping into ATM, and thus no further standardization work is required in this case. Non-defined mappings are flagged in this table with an "R" for "required". For example, mapping over FC is an ongoing project which needs to be supervised by the SMPTE and the EBU, in order to give the broadcasters enough input. Also, DV mapping into ATM is not defined for the moment, but it is needed to bring these signals out of the studio and to effect transfers of these signals between studios.

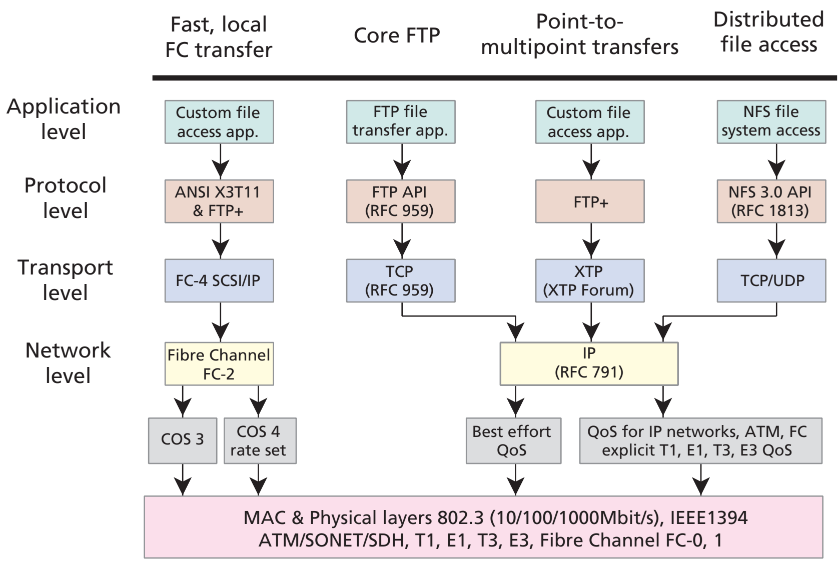

Figure 3
Overview of File Transfer technologies.

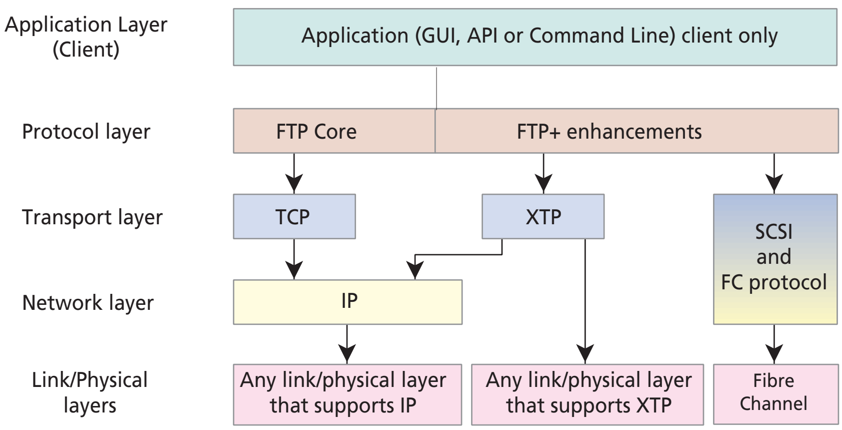

Figure 4
"Special" File Transfer technologies.

3. File transfer

File transfer – the movement of file contents in "push" or "pull" modes over point-to-point and point-to-multipoint topologies – is going to play an increasing role in all the future studio interconnects. It is a guaranteed, error-free, bit-by-bit copy of the content. This requires that the links between the devices participating in the file transfer need to allow for the re-sending of corruptly-received data. This implies bi-directional links.

File transfer allows for different transfer rates:

- slower than real-time;
- real-time (which means that a file containing a 90 minute programme is transferred in 90 minutes);
- faster than real-time.

Why is the last of these so important? Well, if you save time on the transfer, you also save money.

A file transfer is described by a header and after that the content follows. The header is usually sent once only, so that if the file transfer has already started, receivers cannot "join" the transfer in mid-progress.

File transfer is supported for many different types of links (in the physical layer). *Fig. 3* is a rather complex overview of the file transfer technologies we have identified. Very important to note is that there are different types of transport mechanisms which can be used for file transfer – we have ATM, FC, Ethernet, IEEE 1394 (Firewire) and others. However, in order to achieve the interoperability we so much all require, we have defined only a limited number of file transfer protocols. For example, if you are using FC and you have to realize a very, very fast file transfer, then you can use the ANSI standard for FC and the so-called FTP+ protocol. For core (baseline) file exchanges, we recommend the use of FTP as a really basic communications protocol that should be supported by everybody. Distributed file access is only an intermediate solution, simply because it is available at the moment.

The enhanced protocol already mentioned, FTP+ (which has been defined in part by the Task Force), is now the subject of further work within the SMPTE to complete the standardization process. FTP+ uses an FTP baseline protocol (a public standardized protocol) and uses TCP/IP for the underlying transport mechanism. We discovered that FTP (as currently defined) does not meet all our requirements, especially in the case of point-to-multipoint transfers, partial file transfer and, of course, for the very high-speed requirement we have in our applications.

The subgroup has generated the following recommendations for file transfer:

- **FC A/V** for high-performance local-area networks, because we all know that new hard disks will have an FC adapter fitted;
- **ATM** is the choice for wide-area networks.

- **IP-based** interfaces can be used, because IP provides a standardized interface on which our file transfer protocols can sit. So, interoperability is achieved.
- **XTP** is our current choice to meet the requirement for point-to-multi-point file transfer.

4. Conclusions

In conclusion, we have defined a Reference Architecture which allows us to move digital content between the devices of different manufacturers. We have made some choices for general-purpose streaming and for file transfer. We have identified special technologies required to meet our high-end TV broadcast requirements. We have also identified the follow-up work which is necessary to standardize all these new technologies and to expand on the ideas we have already generated. And finally, if every manufacturer and system designer follows just one or two parts of this Reference Architecture, *interoperability* (at least in the physical area, the data-link area and the network area) is guaranteed.

Hans Hoffmann *studied Electronic Communication Engineering in Munich and joined the IRT in 1993 where he developed a jitter measurement device for SDI. Today his work at the IRT is concerned mainly with packetized interfaces (SDTI, Fibre Channel, ATM etc.) and the transfer protocols which will allow the use of packetized bit-rate-reduced signals in a studio environment.*

Since 1993, Mr Hoffmann has been a member of several national and international working groups. He has chaired EBU project groups P/BRRTV and P/PITV, and was very involved in the standardization of the SDTI. He is a member of various SMPTE working groups and contributes to ITU-R Working Group 11B.

Hans Hoffmann chaired the EBU/SMPTE Task Force sub-group on Networks and Transfer Protocols and then, in November 1998, he became Chairman of SMPTE Committee N26 (File Management and Networking Technology).

Coverage of UK DTT service is exceeding all predictions

According to the media consultancy firm, TBS, coverage of the recently-launched UK digital terrestrial TV service is exceeding all expectations. Using a Philips DTX6370 set-top box (already on sale in UK retail outlets), TBS evaluated digital TV reception inside more than 100 buildings in and around London, including residential, business and other types of property.

In the vast majority of cases, existing outdoor antenna systems produced faultless reception of the six DTT multiplexes broadcast from the Crystal Palace transmitting station in South-East London. The only failures occurred where the existing antenna was either very badly damaged or had fallen down (the analogue reception at these locations was already very poor or non-existent).

As far away as Oxford (where analogue reception from Crystal Palace is very noisy – even with a high-gain amplified antenna system), the DTT multiplexes were received perfectly. This proved that DTT signals can be received well beyond the traditional service area of an analogue TV transmitter – even in the presence of strong local analogue TV signals. Domestic-quality TV distribution systems – even very complicated ones – also coped faultlessly with the DTT signals.

Reception of the DTT signals on indoor antennas also exceeded all expectations. Despite the high building "clutter" in London, the signals could be received "ghost-free" at many locations where only a cheap set-top antenna was available. In most cases, the indoor antenna had to be placed close to a window facing Crystal Palace but reception at the more difficult interior locations could usually be achieved by fitting a cheap set-top antenna amplifier.

The COFDM transmission system of DTT, as expected, dealt very effectively with aircraft flutter and antenna masts swaying in the wind – problems which can ruin analogue TV reception near airports or where very tall masts are required to minimize "ghosting" and noise (e.g. in many cities and in remote mountainous areas).

The TBS findings also give high praise to the DTT receiver's "plug and play" capability, and the very useful functionality of its Electronic Programme Guide (Event Service Guide).

TBS can be contacted by telephone on +44 171 286 5570, or by e-mail at: 101722.2700@compuserve.com

Everyone needs standards

D. Wood
Head of New Technology, EBU

In this short article, based on a presentation given to the ABU Technical Assembly in Shanghai, China, the Author argues for a rationalization of the ITU-R in order to strengthen its position as a major standards body in the audio-visual field.

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Manuscript received: 12/11/98.

Introduction

What makes new media delivery technologies successful is not one, but a combination of three elements: *technology*, *infrastructure* and *content*.

The technology needs to be feasible. So, the first question that needs a positive response, if the system is to succeed, is: *"can I make it?"*. This might be a pertinent question for some new technologies such as flat-panel HDTV, but it is often not the biggest barrier to success.

The infrastructure also needs to be possible in the sense of the spectrum space being available. Also, the production equipment, transmission equipment and reception equipment must be available and affordable. So, the second question that needs a positive answer is: *"can I deliver it?"*.

The third area is programme content, which needs to be desirable and sufficiently compelling for the viewer or listener to want it, given that it is available. The third question that needs a positive answer is thus: *"if I supply it, will they want it?"*. The added value of content and technical quality must exceed their cost to the viewer or listener.

Success depends on having all these above things right at once. It is not a "best of three" or "either one or the other". Everything has got to be right at the same moment – the time of the launch – for success to be achieved.

Standards are one of the means to make the technology and infrastructure elements strong. Infrastructure means frequency availability, production equipment availability, transmitter cost and availability and – critically important – receiver cost and availability. It includes the standards for the different elements of the broadcast chain.

Choosing a standard

Given that you need a standard, how should it be chosen? For a nation, or an organization, there are three routes:

- The first is to develop it yourself. This isn't always possible. Maybe your market size is not sufficiently large to support a standard of your own. Possibly, in order to create the standard needs some know-how that you don't have.
- The second route is to choose someone else's own standard. This too has drawbacks. If you choose a system that is proprietary or only used in some places, you could be left trapped with just a small number of suppliers. There could also be diplomatic dimensions to selecting one or other country, or company, for the standard. It is also always difficult to know if the one you have selected is genuinely the best one.
- The third route is to choose a common internationally-agreed standard. This is arguably the best option – you are surer that the standard is a good one, and it may even be the best one, and there are not likely to be supplier problems or political problems.

The case for common standards

Common standards make sense from many directions. They help to create economic efficiency – the best possible goods at the lowest possible prices – and they help competition. In a world of open agreed standards, competition can be on price and features, and not subject to technological barriers. Markets are larger, so production volume can be larger.

There are many cases where open standards have been the making of an industry. We could include here the ISA IBM-compatible bus in personal computers, the GSM telephone, and many more.

No one can claim that the use of open standards is the guarantee so success – it certainly is not. All the infrastructure elements and – most important – the content elements are needed too. But at least with an open standard, you minimize many of the barriers to success.

In the recent WBU-TC handbook on digital radio, the editor notes with regret that there are currently 11 different digital radio systems being used, or being defined, in different parts of the

world. He contrasts this with the world of analogue radio, where there are just two standards in use throughout the world. Is this really progress?

Reasons for multiple standards

Why do multiple standards arise? There is a catalogue of reasons.

The development of new systems can be done at different times, and the developers have thus different technical toolkits to use, because of technical evolution. Even a few months can make a difference.

The developments can be done by different groups of people. Give a team of engineers the choice of either using someone else's system, or spending money to develop their own, and most will go for the latter.

It can also be that needs are different in different organizations, and different parts of the world. What the system is required to do may be different. There may also be differences in infrastructure – such things as transmitter density or population density. They may cause different priorities. It is also not unknown for some organizations to see a business advantage in having their own system.

Agreeing common standards is inevitably difficult. But even if it isn't always possible, a few successes can be well worth the energy spent.

Arranging for standards agreement

How can we help common world-wide standards to happen? There may be four principle ways to do so.

- the first, and main one, is to arrange effective standards organizations;
- the second is to disseminate information and encourage innovation throughout the world, to prevent time-differential development;
- the third is to start early with the standards process and to give it all the energy possible, before positions become too entrenched;
- the fourth is to be fair, in terms of technology and economic advantage.

All of the world's technical standards should not come, for example, from one country or region. We need a balance, so that every organization is encouraged to continue in the pursuit of world-wide standards. They must feel that any efforts they put in to the development of standards are not a waste of time, because of a dominant actor.

Standards bodies

If we look at the world media standards bodies, and put aside for the moment the regional bodies, we find two kinds. There are *official* standards bodies and *private* bodies. In the official category come the ITU-R, ITU-T, IEC TC 100 and the ISO / IEC JTC1. In the category of private bodies come DAVIC, DVB (well, almost world-wide), DRM, the W3C, the IETF and others.

Both types of bodies have had their successes, and lack of success. In the ISO / IEC JTC1, to cite an example from the official sector, we have seen remarkable world-wide agreement on JPEG (compression for still pictures), MHEG (multimedia content decoder), and MPEG (video compression, now also Metadata). In the private sector, we can cite as examples of success, all the WWW standards developed in the IETF.

As an example of the particular dilemma that private bodies face, there are two ways of working that can be employed. The first is to keep technical details to yourself during the development via a non-disclosure agreement. This was done, for example, in the HD-MAC project and is now being done in the DRM project. There are advantages to this way of working, particularly the clarity of patent rights when any money comes to be made.

The other way is to keep the technical details open, as you develop the system. This was and is done for example in the DVB, MPEG, MHEG, JPEG and DAVIC projects. It has the advantage that there is a greater chance that people outside the project will buy in to it, if they have been able to follow it, and understand why technical choices are being made.

NOTE: The use of the above logos is for illustrative purposes only. It does not imply any endorsement by these standards bodies of any of the views presented in this article.

1. The ITU standards activities

In principle, the ITU-R should score heavily as a standards body. Its very essence is the need for solidarity of all nations, rich and poor, large and small. Furthermore, it has highly trained and competent staff. It also has great technical communications infrastructure and facilities. It should be the focus of the world's media delivery standards. However, in spite of this promise, the attendance at the ITU-R is diminishing, and other standards bodies – both official and private – are increasingly drawing larger crowds and are agreeing matters relating to broadcasting.

Why should there be a drift away from the ITU to other bodies? The reasons given by delegates to other bodies include:

- it is no longer possible to agree unique common standards in the ITU-R;
- the ITU-R simply catalogues other people's multiple standards;
- some also say that the ITU-R is dominated by administrations;
- others say that the procedures are too slow.

But what determines if a standards body is successful? Can we use this to instigate steps to give the ITU-R back its pole position? A standards body needs good leadership quality, good secretariat quality, breadth of membership, and a structure that matches the environment.

What steps should be taken? We probably need to make the structure fit the current world of media delivery better. We need to get the best chairmen around. If we do that, there is a chance that, like a "field of dreams", if we build, then they will come.

The media delivery environment

What is the context of the media delivery environment that needs to be matched?

Perhaps there are two important specific trends. The first is that baseband standards will eventually become software platform standards. The second is that radio frequency standards will be agile standards, able to respond to different parameter sets by means of broadcast configuration information.

However, the real dominant trend today is "convergence". This is the coming together of all media, with common generic systems. This is the fundamental change ahead of us. What is transported will not specifically be pictures, sound and data – it will be multimedia which includes sound, vision and other elements. What is more, this will need to be transported in an interchangeable manner by all delivery systems. If there is a fundamental concept for the ITU to respond to, it is convergence.

The EBU has offered some suggestions to the ITU-R about how to arrange the studies in future. These suggestions are as follows:

- ITU-R Study Groups 10 (sound) and 11 (Television) should be merged to take into account the future multimedia transport environment. Sound and Television will always have individual identities, and permanent separate futures at the level of content. However, at the technical level, they need to have compatible coding and transport systems.
- The combined Study Group should be arranged in a horizontal structure with horizontal generic slices that cover all delivery platforms. These might be groups on (all aspects of) studio systems, coding and multiplexing, radio frequency systems, and frequency planning. The ISO/IEC JTC1 made the same suggestion at their last meeting, in an effort to encourage co-operation with the ITU-R in future.

Conclusions

Common standards help everyone. In order to prepare them, active efficient standards bodies are needed. In the media standards world, the ITU is ideally placed to be the pivotal body, but some changes are needed to align it with technological trends - principally *convergence*.

Any change is painful, but it will be worth it. The media delivery horizon is exciting, to say the least. We can look forward to systems for broadcasting multimedia to mobiles, for a range of synergies between the Internet and Broadcasting, to systems which use client storage in the home receiver, and systems which use the Internet to deliver television and radio broadcasts.

If we can move towards greater common standards in these areas, we will certainly have the gratitude of our grandchildren. Let us try.

EBU Website – useful technical pages

If you haven't already done so, why don't you bookmark the following useful EBU technical pages in your Internet brower:

EBU Technical home page:	http://www.ebu.ch/technical.html
EBU Technical Committee:	http://www.ebu.ch/tech_tc.html
Broadcast Technology:	http://www.ebu.ch/bmc_index.htm
Production Technology:	http://www.ebu.ch/pmc_home.html
Network Development:	http://www.ebu.ch/ndc_home.html

New DVB standard for DSNG
– and contribution satellite links

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In July 1997, the Technical Module of the DVB Project set up an Ad-hoc Group on Digital Satellite News Gathering (DSNG) under the chairmanship of the RAI. Its tasks were: (i) to define the specification of the modulation/channel coding for DSNG and other contribution applications by satellite, (ii) to define the specification for the auxiliary co-ordination channels and (iii) to co-operate with other DVB groups in order to define the user guidelines for source coding, Service Information (SI) and scrambling for Conditional Access (CA).

A flexible DVB-DSNG system [1] has now been defined and is described here. Mainly based on the DVB system for satellite broadcasting (DVB-S), it offers a range of different picture-quality levels at various bit-rates by using the MPEG-2 MP@ML and 422P@ML algorithms. The specification for the auxiliary co-ordination channels [2] was finalized by the group in Autumn 1998 and the approval procedure is still in progress within DVB and ETSI. It is not described in this article for reasons of conciseness.

1. Introduction

Today's broadcasting is dominated by increasing competition and, consequently, the real-time acquisition of news material (e.g. sports events, interviews, concerts, calamities) – in both the domestic and the international environments – is a major factor in the search for audience ratings. In this context, light-weight satellite news-gathering (SNG) transmit terminals – with 90 to 150 cm antennas – offer a cost-effective solution to establishing rapid connections between outside broadcast (OB) vans and TV studios without requiring local access to the fixed telecom network.

In the case of PAL, SECAM and NTSC, *analogue* SNG systems using frequency modulation (FM) are currently operated in both the C and Ku bands. In Europe, the TV satellite contribution links are commonly in the Ku band (using the 14 - 14.5 GHz band for the up-link and the 10.71 - 12.75 GHz band for the down-link). Although there have been progressive improvements in antenna and amplifier design, and the original bulky and heavy analogue SNG equipment has been reduced in weight and size, portability is still very much a key issue requiring adequate solutions in the analogue world.

The required EIRP of an up-link of course depends on the footprint of the satellite which will receive its signals. The EIRP of analogue up-links are typically 69 - 75 dBW, depending on the size of the antenna and the high-power amplifier (HPA) in use. Antenna sizes vary from 1.5 to 2.4 m and the HPA powers range from 300 to 600 W.

The commercial introduction of small digital equipment in the domains of video/sound compression, advanced error protection and modulation has recently enabled the development of operational *digital* SNG (DSNG) systems. These new systems have a number of advantages over analogue systems, including:

- *miniaturization* of the up-link terminal;
- lower required EIRPs;
- more efficient use of the frequency spectrum.

Digital SNG systems permit multiple signals to be transmitted simultaneously through the satellite transpond-

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ers, significantly increasing the flexibility of the transponder access, and reducing the cost per channel. The inherent flexibility of DSNG systems allows us to fulfil the different quality requirements of news, sports events and entertainment by operating the video / audio compression algorithm at the most appropriate bit-rate. Moreover, the ruggedness of a digital system against noise and interference enables constant picture and sound quality to be obtained at the receiving site, down to a certain threshold signal level.

Before the development of the DVB-DSNG standard, digital contribution links by satellite were often based on the ETSI 300 174 standard for video compression [3]. This system, which was designed for *contribution* applications at 34 and 45 Mbit / s, also became available in proprietary scaled versions at 17 Mbit / s and 8.5 Mbit / s which were more suitable for pure DSNG applications. The modulation and channel coding were usually based on the IDR specification, using QPSK modulation and convolutional coding.

In 1993-94, the DVB Project developed the specification of a digital multi-programme television system for satellite broadcasting (DVB-S), under the direct responsibility of the RAI Research Centre [4][5]. With the world-wide success of this system, it became more and more clear that it could be suitable also for DSNG applications, with significant advantages in cost, performance and flexibility over the previous systems [6]. In the summer of 1997, the DVB Project decided to start the development of a new specification for DSNG,

Abbreviations

422P@ML (MPEG-2) 4:2:2 Profile at Main Level
8PSK Eight-phase-shift keying
16-QAM 16-state quadrature amplitude modulation
64-QAM 64-state quadrature amplitude modulation
ATM Asynchronous transfer mode
AWGN Additive white Gaussian noise
BER Bit error rate
C/I Carrier-to-interference ratio
C/N Carrier-to-noise ratio
CA Conditional access
CATV Community antenna television
CBPS Coded bit per symbol
CM (DVB) Commercial Module
DCT Discrete cosine transform
DPCM Differential pulse code modulation
DSNG Digital satellite news gathering
DTH Direct-to-home
DVB Digital Video Broadcasting
DVB-S DVB - Satellite
E_b/N_o Ratio between energy-per-useful-bit and the spectral density of the noise
EIRP Effective isotropic radiated power
EPG Electronic programme guide
ETSI European Telecommunication Standards Institute
FDM Frequency division multiplex
FDMA Frequency division multiple access
FEC Forward error correction
G/T Gain/temperature ratio
GoP Group of pictures
HPA High power amplifier
IBO Input back-off
IDR Intermediate data-rate
IEC International Electrotechnical Commission
IMUX Input multiplexer
IPFD Isotropic power flux density
IRD Integrated receiver/decoder
ISI Inter-symbol interference
ISO International Organization for Standardization
ITU International Telecommunication Union
ITU-R International Telecommunication Union, Radiocommunication Sector
LNB Low-noise block
MCPC Multiple channels per carrier
MMDS Multipoint microwave distribution system
MP@ML (MPEG-2) Main Profile at Main Level
MPEG (ISO/IEC) Moving Picture Experts Group
OBO Output back-off
OMUX Output multiplexer
PAT (MPEG) Programme Associated Table
PDH Plesiochronous digital hierarchy
PMT (MPEG) Programme Map Table
PRBS Pseudo-random binary sequence
PSI (DVB) Programme Service Information
QEF Quasi-error-free
QPSK Quadrature (quaternary) phase-shift keying
R-S Reed-Solomon
RAI *Radiotelevisione Italiana*
SCPC Single channel per carrier
SDH Synchronous digital hierarchy
SI (DVB) Service Information
SNG Satellite news gathering
TCM Trellis-coded modulation
TSDT (MPEG) Transport Stream Descriptor Table
TS (MPEG) Transport Stream
TWTA Travelling-wave-tube amplifier
VLSI Very large-scale integration
XPD Cross-polar antenna discrimination

based on the DVB-S system [4] but with a number of new features designed to cover the commercial and operational requirements of contribution applications. This project was also the responsibility of the RAI.

The DVB-DSNG system [1] is transparent to any signal in the MPEG-2 Transport Stream format and can transport video signals encoded in either the MP@ML format, or in the 422P@ML format when higher quality and enhanced editing facilities are required. Other MPEG-2 profiles and levels may be transported as well, e.g. 422P@HL which is suitable for HDTV contribution links.

The DVB-DSNG system is based on QPSK modulation and convolutional coding, which were originally developed to provide DTH television services via satellite in the MCPC mode, but which are also suitable for DSNG and contribution applications in the SCPC mode. Nevertheless, two optional transmission modes have been added to the DVB-DSNG system: trellis-coded 8PSK and 16QAM. These offer higher spectrum efficiency in those applications which are less affected by power limitations (e.g. vehicle-based DSNG up-links). The main feature of the DVB-DSNG system is thus the flexibility of its modulation and channel-coding schemes which – on a case by case basis – allow the most appropriate modulation scheme, symbol rate and coding rate to be selected in order to optimize the satellite link performance (i.e. the spectral occupancy within the satellite transponder and the power requirements).

Specific technical solutions have been defined by DVB for the transport of MPEG signals on terrestrial telecom networks such as PDH and SDH, by mapping the Transport Stream packets into ATM cells. These adapters can be used to connect the DSNG receiving stations to the TV studios.

2. Basic user requirements

The technical characteristics of DVB systems are largely market-driven. Based on an analysis of the market needs, the DVB Commercial Module (CM) produces *Commercial Users' Requirements* for input to the DVB Technical Module (TM) which is responsible for developing the required specifications.

In accordance with the ITU, the CM has adopted the following definition of SNG (ITU-R Rec. SNG.770-1): *"Temporary and occasional transmissions with short notice of television or sound for broadcasting purposes, using highly portable or transportable up-link earth stations operating in the framework of the Fixed-Satellite Service (FSS)"*.

A DSNG "terminal" or "up-link" is a portable (or transportable) earthstation which can be moved to a remote location in order to transmit back the video programme with its associated sound, or sound-only programme signals, either "off-tape" or "live". It can either be packaged in "fly-away" form (i.e. in flight cases suitable for air transportation) or integrated into a vehicle.

DSNG up-link terminals should be highly reliable and have reduced size and weight, while the receiving station should be dimensioned appropriately to ensure the required link availability. Therefore the transmission format should provide both high ruggedness against noise and interference and the best exploitation of satellite capacity.

High intervention promptness and low set-up complexity are required. In particular, *"the equipment should be capable of being set up and operated by a crew of no more than two people within a reasonably short time (for example, one hour)"*. Interoperability between different pieces of equipment is another key feature of DSNG, especially in an international programme-exchange environment. In particular, the CM has identified within the complex DVB / MPEG SI / PSI tables a possible source of problems for DSNG, affecting equipment interoperability and rapid link set-up.

By nature, DSNG links are contribution links, the quality objectives of which are defined by ITU-R Rec. BT.1121: *"There is no need to define lower quality objectives, if it is understood that, due to circumstances, possible relaxations are to be accepted by the user. For DSNG links, the typical bit-rate used by fly-away and small transportable terminals are about 8 Mbit/s, using MPEG-2 MP@ML. However for transportable stations"*, when higher quality and enhanced editing facilities are required, *"use of MPEG-2 422P@ML should be supported. … In this case, bit-rates should be higher than 8 Mbit/s and lower than 34 Mbit/s"*.

As regards multiplexing, although DSNG transmissions usually transport a single TV programme and its associated sound signals (SCPC), *"advantage should be taken of the flexibility of the MPEG-DVB multiplex"* to convey multiple programmes (MCPC).

The processing delays of digital compression systems may be very high (even exceeding one second), especially with today's sophisticated coding algorithms which allow high bit-rate compression ratios. Short video-coding delays are important characteristics for those applications where the DSNG transmission is mixed together with a live programme, since long delays would prevent dialogues between journalists in the studio and in the field.

Optionally, DSNG equipment should be capable of providing two or more duplex co-ordination (communication) circuits by satellite, whenever possible in the same transponder as the main DSNG signal. These channels should be available prior to, during and after the DSNG transmission in order to connect the DSNG operator, the satellite operator and the broadcaster. This equipment may also provide for data transmission and faxes. The specification [2] was finalized within the DSNG group in Autumn 1998 and the approval procedure by DVB and ETSI is now in progress.

Regarding the equipment cost, the CM pointed out that *"the total cost of the system and its operation should be considered, and not just the receiver cost. A non-negligible part of the overall cost of an SNG transmission lies in the requirements for satellite capacity. Modulation techniques, additional to QPSK, such as 8PSK and 16QAM, should be investigated to optimize the efficient use of satellite capacity"*.

3. Source coding and multiplexing

The success of the DVB standards is also due to the adoption on all media (e.g. satellite, CATV, terrestrial VHF/UHF and MMDS networks) of a "common solution" for video / audio coding and digital multiplexing, making possible the mass production of VLSI chips for consumer IRDs.

3.1. Video coding

The MPEG-2 MP@ML format may be used as the baseline solution for picture coding in DSNG applications. It allows high flexibility, being able to operate with variable bit-rates from 1.5 to 15 Mbit/s.

MPEG-2 codecs are based on Hybrid DPCM/DCT algorithms with motion compensation, operating on I-frames (intra), P-frames (predicted) and B-frames (bi-directional prediction). It should be noted that MP@ML is a 4:2:0 system which was designed for distribution rather than contribution. At bit-rates of 6 Mbit/s and 9 Mbit/s, it allows a subjective quality for current programme material that, respectively, is equivalent to PAL and 4:2:2 pictures. Lower bit-rates may be acceptable for specific applications (e.g. films, news, educational), where power and bandwidth limitations are dominant over the picture quality requirements.

In 1995, MPEG-2 defined a picture coding "profile" called 422P@ML to fulfil the requirements of the production environment. It offers a number of additional features compared with the MP@ML format such as (i) the coding rate can be increased up to 50 Mbit/s and (ii) the chroma components maintain the same 4:2:2 format as the uncompressed studio format. This allows: higher picture quality; better chroma resolution; post-processing after co-decoding; and short GoPs to improve the editability in compressed form and to shorten the coding delay. Subjective quality tests (non-expert viewers, 4H distance) have been carried out by the RAI Research Centre and other organizations [7] on computer-simulated 422P@ML sequences, with single and multiple generations (eight co-decoding processes) and colour matte post-processing.

Different GoP structures have been analyzed:

➩ a purely intra-frame configuration which allows one-frame editing precision at the expense of low compression efficiency, running at 50 and 30 Mbit/s (indicated as *I@50* and *I@30);*

➩ one I-frame and one B-frame, which allows a good compromise between editing and compression ratio, running at 30 and 20 Mbit/s (indicated as *IB@30* and *IB@20*);

➩ the traditional MP@ML GoP, with 15 IBBP frames, running at 20 Mbit/s (indicated as *IBBP@20*).

With reference to the double-stimulus continuous quality 100-grade scale, the following quality levels are arbitrarily defined in MPEG documents:

➩ *"transparent"* (0 to 12.5);

➩ *"nearly-transparent"* (12.5 to 20);

➩ *"good"* (20 to 40).

The subjective test results indicate that after eight co-decoding processes, the *I@50* (including chromakey) and *IB@30* coding structures fulfil the "transparency" ratings and, after a single co-decoding process, chromakey can be performed "transparently" on *I@30* and *IB@20*. In addition, all the coding structures gave ratings which matched well the "nearly transparent" quality, apart from a few tests which moderately exceeded the 20% target.

Summarizing, to fulfil the wide range of picture-quality levels and bit-rates required by DSNG and other contribution applications, MP@ML at bit-rates from 1.5 to 15 Mbit/s can cover the applications where no (or very limited) post-processing is performed in the studio before re-broadcasting. The 422P@ML format at bit-rates from 15 to 30 Mbit/s can, on the other hand, cover high-quality applications where post-production and cascaded co-decoding are required.

In any case, it must be kept in mind that the switching and editing of MPEG-2 transport streams in the studio, without decoding, can be very difficult because of the problems related to clock handling and buffer overflow control. Therefore, in many cases the DSNG contributions (independently of the adopted compression scheme) must be re-converted into 4:2:2 format in the studio, edited and then re-encoded for final broadcasting (in MP@ML format).

3.2. Audio coding

All the DVB systems – in line with the trend toward international standardization – have adopted the MPEG Layer II audio coding method which allows a wide range of bit-rates (e.g. from 64 to 256 kbit/s) in order to satisfy the various service requirements. Bit-rates as low as 64 kbit/s may be applicable for some DSNG applications with mono channels. The optional use of linear (uncompressed) audio coding is also under evaluation by DVB for contribution applications that require maximum audio quality.

4. Transport Multiplexing and Service Information (SI)

The DVB-S system adopts a common framing structure, based on the MPEG-2 transport multiplex, with fixed length packets of 188 bytes, comprising one sync byte, three header bytes and 184 useful bytes. This structure allows easy interworking between broadcast channels and telecom networks using ATM protocols.

The MPEG-2 multiplex is very flexible for merging a variety of video, sound and data services in the transport stream, as well as additional information (e.g. Service Information, Conditional Access). Therefore it allows SCPC as well as MCPC services.

The DVB-MPEG Service Information tables defined for broadcasting applications describe in detail the multiplex configuration and the programme content, and allow the user to access easily a wide programme offer through the Electronic Programme Guide (EPG). Annex D of the DSNG specification deals with a simplified Service Information mechanism based on few fixed tables. It avoids the need to compile SI information in the field, thereby accelerating the link set-up time and simplifying any interoperability problems. Up-link station identification is also provided, for emergency interference situations.

Of the MPEG2-defined SI tables that have been introduced to describe the multiplex configuration and the programme content in broadcasting services, only the Programme Associated Table (PAT), the Programme Map Table (PMT) and the Transport Stream Descriptor Table (TSDT) are kept in the DSNG specification as mandatory. These tables may be fixed for a DSNG station. In the TSDT table, a descriptor indicates that the Transport Stream is for Contribution Applications (not for the general public). Furthermore, for DSNG transmissions, another descriptor is inserted to allow fast identification of the up-link station in cases of transmission problems (e.g. access to the wrong transponder or satellite).

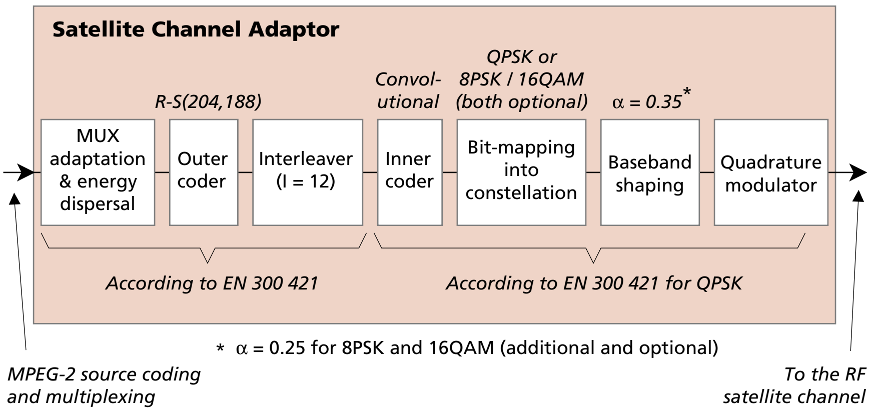

Figure 1
Functional block diagram of the DVB-DSNG system.

These SI structures may prevent compatibility with consumer IRDs and, therefore, if this compatibility is required by the operator, all the SI tables must be compiled according to the DVB-SI specification.

Since no forward error correction (FEC) protects the TS packet headers, a rugged "channel adapter" is required to provide an *error-free* datastream to the demultiplexer input, as described in the following section.

5. Channel coding and modulation

The transmission performance of a typical DSNG system depends on the various components included in the satellite chain:

- the transmit earthstation;
- the space segment (up-links and down-links);
- the satellite transponder (IMUX, OMUX filters, TWTA);
- the receive earthstation.

The satellite channel is basically *non-linear, wide-band* and *power limited*. The main signal impairments are introduced by noise, rain attenuation and interference on the space segment, and eventually by incorrect alignment of the transmit and receive stations and equipment. The non-linearity (amplitude and phase distortions) of the on-board TWTA is responsible for impairments to the overall system performance.

In the case of digital DTH services, a single QPSK carrier is transmitted in the transponder and, to achieve the maximum power efficiency, the satellite TWTA is usually operated close to saturation. The effects of TWTA non-linearity are waveform distortion and side-lobe regeneration of the power spectrum. In these applications, due to the reduced dimensions of the receiving antennas, the service availability is mainly limited by the down-link noise.

For DSNG and contribution applications, the usual method of accessing the transponders is FDM where part of the transponder bandwidth (frequency slot) is allocated to each signal, in SCPC mode. In order to reduce the effect of intermodulation noise introduced on adjacent carriers occupying the same transponder, the TWTA must be operated significantly below the saturation point. The linearity requirements are raised also by the fact that the aggregated FDM signal is no longer characterized by a constant envelope, even if each individual signal has a quasi-constant envelope (e.g. QPSK or 8PSK). The higher the spectrum efficiency of the modulation/coding scheme, the more stringent are the linearity requirements, because of the reduction of the system ruggedness against intermodulation interference from the adjacent signals.

In FDM transmissions, the down-link level of any one signal does not noticeably change as a function of the total transponder load, making it straightforward to set and monitor the down-link EIRP. This more linear type of operation also provides more protection against single up-link drive fluctuations.

Efficient and reliable transmission of digital television signals over satellite channels is focused on the design of the "channel adapter", which performs the adaptation of the multiplexed video/audio/data bitstream to the physical channel, by adopting powerful channel coding and modulation techniques. In the definition of the DSNG system, the design target has been the minimization of the effects of the various channel impairments, such as additive noise, interference from analogue and digital signals, and linear and non-linear distortion. The specified system offers many transmission modes (inner coding and modulations), giving different trade-offs between power and spectrum efficiency. QPSK modulation has been adopted (and optionally the 8PSK and 16QAM modulation schemes) and the concatenation of convolutional and Reed-Solomon codes is introduced.

The QPSK mode is compliant with the DVB-S system defined in [4] while for 8PSK and 16QAM, "pragmatic" trellis coding [8] has been applied, optimizing the error protection of the same convolutional code. The convolutional code is able to be configured flexibly, allowing the optimization of the system performance for a given satellite transponder bandwidth. The QPSK and 8PSK modes, thanks to their quasi-constant envelopes, are suitable for operation with saturated satellite power amplifiers, in a single-carrier-per-transponder configuration. 16QAM (as well as QPSK and 8PSK) is appropriate for operation in quasi-linear satellite channels, in multi-carrier FDM-type applications, where better spectrum efficiency is attained.

Fig. 1 gives a functional block diagram of the DVB-DSNG transmission system. The input stream, organized in 188-byte packets following the MPEG-2 transport multiplexer [9], is randomized bit by bit through a scrambling PRBS, in order to comply with the Radio Regulations interference requirements (the transmitted signal must have a regular spectrum shape), and also to facilitate clock recovery in the receiver. Then the shortened R-S code (204,188, t = 8) – derived from the original R-S (255, 239, t = 8) code – is applied to each randomized transport packet.

Since, on the receiver side, the residual errors at the output of the Viterbi decoder are not statistically independent, but are grouped in a burst which may overload the R-S code correction capability, the error distribution is randomized through a convolutional interleaver with depth I equal to 12

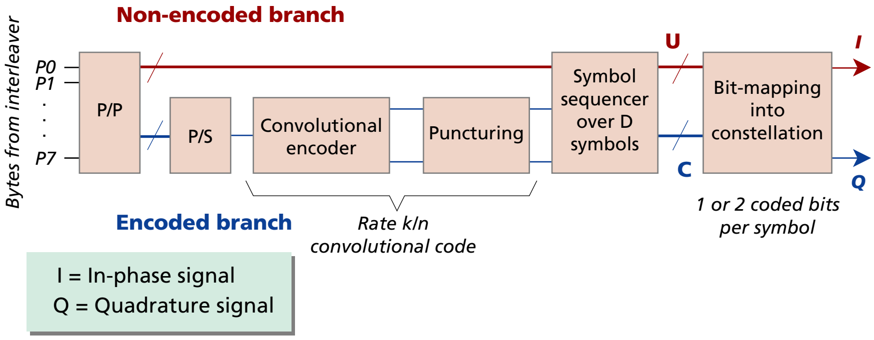

Figure 2
Principle of the inner trellis coder.

bytes applied to the error-protected packets. The interleaved packets are then passed to the convolutional encoder, which is based on a rate 1/2 mother convolutional code with constraint length equal to 7 (64 trellis states), and which allows the selection of the most appropriate level of error correction for a given service or data-rate.

Punctured convolutional coding is associated with QPSK modulation (according to the DVB-S system specification [4]) with the possibility of operating at five possible rates: 1/2, 2/3, 3/4, 5/6, 7/8. Pragmatic Trellis Coded Modulation (TCM) [8] on the other hand is associated with 8PSK and 16QAM. The operating principle of the pragmatic trellis encoder is shown in *Fig. 2*.

The byte-parallel stream at the output of the convolutional interleaver is conveyed to a parallel-to-parallel (P/P) converter which splits the input bits into two branches, depending on the selected modulation / inner coding mode. It has been designed to reduce, on average, the byte error-ratio at the input of the R-S decoder (high concentration of bit-errors in bytes) and, therefore, to reduce the bit error ratio (BER) after the R-S decoder.

In the *non-encoded* branch, the symbol sequencer generates a sequence of signals **U**, each to be transmitted in a modulated symbol. These bits generate parallel transitions in the trellis code, and are only protected by a large Euclidean distance in the signal space. In the *encoded* branch, the signals first pass through a parallel-to-serial (P/S) converter for subsequent processing by the punctured convolutional encoder. These bits generate, through the symbol sequencer, a sequence of signals **C**, each to be transmitted in a modulated symbol.

The 8PSK 5/6 and 8/9 schemes are characterized by one coded bit per symbol (referred to as 1CBPS), while the 8PSK 2/3 and 16QAM 3/4 and 7/8 schemes have two coded bits per symbol (2CBPS). The optimum bit-mapping into constellations are different for 1CBPS and 2CBPS. The selection of the trellis coding schemes, from a number of different proposals, was based on accurate computer simulations carried out by the RAI Research Centre.

The selected schemes were the ones which offer the best performance on a linear channel affected by Additive White Gaussian Noise (AWGN). With comparable performance in mind, the 1CBPS schemes are preferred since they require a lower processing speed in the TCM Viterbi decoder, compared with 2CBPS schemes, and therefore they allow the implementation of higher-speed modems (for high-quality contribution applications or MCPC transmissions).

Finally, baseband shaping and quadrature modulation are applied to the signal. Square-root raised-cosine baseband shaping, with a roll-off factor of $\alpha = 0.35$, is considered for all constellations, as in the DVB-S system [4]. An additional roll-off factor $\alpha = 0.25$ may be used for the 8PSK and 16QAM modulation schemes, in order to increase the spectrum efficiency within the transponder bandwidth. This choice was based on extensive computer simulations, including satellite TWTA effects, carried out by the RAI.

Modulation	Inner code rate	Spectral efficiency (bits/symbol)	Modem implementation margin [dB]	Required E_b/N_0 [dB] for BER = 2×10^{-4} before R-S
QPSK	1/2	0.92	0.8	4.5
	2/3	1.23	0.8	5.0
	3/4	1.38	0.8	5.5
	5/6	1.53	0.8	6.0
	7/8	1.61	0.8	6.4
8PSK (optional)	2/3	1.84	1.0	6.9
	5/6	2.30	1.4	8.9
	8/9	2.46	1.5	9.4
16QAM (optional)	3/4	2.76	1.5	9.0
	7/8	3.22	2.1	10.7

Table 1
IF-Loop performance of the DVB-DSNG system.

6. Performance on the AWGN channel

Sensitivity to transmission noise is expressed, for the various rates of the convolutional code, by the E_b/N_0 ratio required to achieve a target residual BER. E_b is the energy per useful bit and N_0 is the spectral density of the AWGN. The DVB-DSNG system has been designed to provide a QEF quality target, i.e. approximately less than one incorrect error event per transmission hour at the input of the MPEG-2 demultiplexer. This target, achievable by interleaving and by R-S error correction, corresponds to a BER of about 2×10^{-4} at the output of the TCM Viterbi decoder and to a byte error ratio of between 7×10^{-4} and 2×10^{-3} depending on the coding scheme.

It should be noted that these evaluations take into account stationary noise only, and ideal demodulation. Furthermore, the effects of phase noise and carrier-recovery instabilities might generate bursts of uncorrectable errors, separated by large time intervals. DVB-DSNG coding schemes are not rotationally invariant (i) because, in the majority of cases, pragmatic schemes were not available and (ii) in order to optimize the BER performance. Therefore, care should be taken in the design of the frequency converters and the carrier-recovery systems, in order to avoid "cycle skipping" and "phase snaps" which may produce service interruptions. These goals may be achieved easily in professional front-ends

Table 1 gives the IF Loop system performance requirements for the different modes, in terms of the required E_b/N_0 to provide BER = 2×10^{-4} (Quasi Error Free quality target). The figures for E_b/N_0 are with reference to the useful bit-rate R_u (188-byte format, before R-S coding), and take into account the factor 10 Log 188/204 ≅ 0.36 dB due to the R-S outer code and the modem implementation margins. For QPSK, the figures are derived from [4]. For 8PSK and 16QAM, modem implementation margins which increase with the spectrum efficiency are adopted, to cope with the larger sensitivity associated with these schemes.

The ruggedness against noise of digital TV (QPSK-3/4) and analogue PAL/FM on a satellite channel are shown in

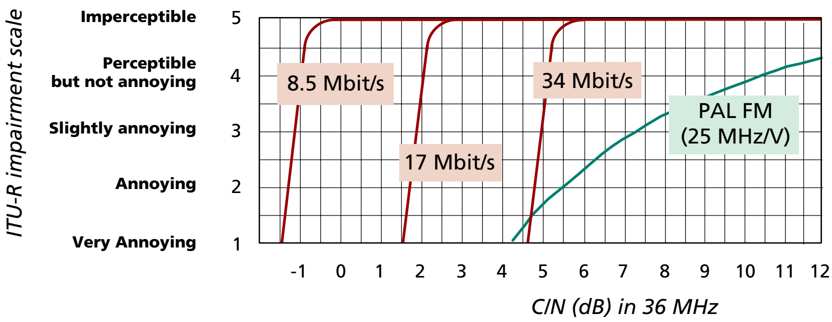

Figure 3
Picture impairment vs. C/N: digital TV (QPSK-3/4) and analogue FM/TV on a satellite channel.

Fig. 3. The quality impairment is expressed in terms of the C/N ratio, assuming as reference an analogue receiver bandwidth B_{RX} of 36 MHz, which is typical of satellite FM/TV transmissions with 25 MHz/V frequency deviation. To perform a fair comparison, the digital system is operated in single-signal-per-transponder configuration, and the C/N ratio is measured in the same bandwidth B_{RX} of 36 MHz as the analogue signal (about 1 dB additional degradation on the transponder should be considered):

$$C/N\ (dB) = E_b/N_0\ (dB) + 10\ \mathrm{Log}\ (R_u/\ B_{RX})$$

From *Fig. 3*, it can be seen that a DSNG signal at 17 Mbit/s, providing near-contribution quality, would require about 3 dB C/N (in 36 MHz) to operate quasi-error-free compared with the 12 - 13 dB required by analogue FM/PAL for an acceptable picture quality. If the transmission rate is reduced to 8.5 Mbit/s, which is suitable for DSNG applications with PAL quality, the required C/N ratio would approach 0 dB.

Thanks to this remarkable performance, the digital solution is capable of delivering the picture and sound quality of the "compressed" source, provided that adequate margin against rain attenuation is allowed for in the link budget design to ensure that the system operates above the service continuity threshold.

7. Examples of the system in use

One of the main features of the DVB-DSNG system is its flexibility. It allows, on a case-by-case basis, the selection of the modulation scheme, the symbol rate and the coding rate in order to optimize the satellite link performance (i.e. the spectral occupancy of the satellite transponder and the power requirements). On the other hand, in order to achieve rapid interoperability and link set-up in emergency situations, the DSNG specification mandates that at least one "user-definable" set-up is available in DSNG equipment. This set-up includes the video/audio coding parameters, the modulation scheme and the symbol rate.

Although DSNG applications usually exploit the satellite bandwidth in the FDM configuration, the DSNG system is suitable also for single-carrier-per-transponder transmissions. In this type of configuration, the transmission symbol rate R_S can be matched to the given transponder bandwidth BW (at –3 dB), to achieve the maximum transmission capacity compatible with the acceptable signal degradation due to transponder bandwidth limitations. To take into account possible thermal and ageing instabilities, reference can be made to the frequency response mask of the transponder.

In multi-carrier FDM configurations, R_S can be matched to the frequency slot BS allocated to the service by the frequency plan, in order to optimize the transmission capacity while keeping the mutual interference between adjacent carriers at an acceptable level.

Fig. 4 gives examples of the maximum useful bit-rate capacity R_u achievable by the system, versus the allocated bandwidths BW or BS. R_u is the useful bit-rate (188-byte format) after MPEG-2 MUX while R_S (symbol rate) corresponds to the –3 dB bandwidth of the modulated signal. $R_S(1 + \alpha)$ corresponds to the theoretical total signal bandwidth after the modulator. The figures for very low and very high

Satellite BW (at –3 dB)	System mode	Symbol Rate R_S [Mbaud]	Bit-rate R_u (after MUX) [Mbit/s]	E_b/N_0 (specification) [dB]
36	QPSK 3/4	27.500	38.015	5.5
36	8PSK 2/3	27.500	50.686	6.9

Table 2
Examples of System configurations by satellite: single-carrier-per-transponder.

Satellite BW [MHz]	Slot BS [MHz]	Number of Slots in BW	Video Coding	System mode	Symbol Rate [Mbaud]	BS/R_S [Hz/Baud]	Bit-rate R_u [Mbit/s]	E_b/N_0 [dB] (specification)
36	9	4	MP@ML	QPSK 3/4	6.1113	1.47	8.4480	5.5
36	18	2	422P@ML	QPSK 7/8	13.3332	1.35	21.5030	6.4
36	12	3	422P@ML	8PSK 5/6	9.3332	1.28	21.5030	8.9
36	9	4	422P@ML	16QAM 7/8	6.6666	1.35	21.5030	10.7

Table 3
Examples of system configurations by satellite: multi-carrier FDM transmissions, SCPC mode.

bit-rates may be irrelevant for specific applications. In these examples the adopted BW/R_S or BS/R_S ratios are $\eta = 1 + \alpha = 1.35$ where α is the roll-off factor of the modulation. This choice allows us to obtain a negligible E_b/N_0 degradation due to transponder bandwidth limitations and adjacent channel interference on a linear channel. Higher bit-rates can be achieved with the narrow roll-off factor $\alpha = 0.25$ (optional for 8PSK and 16QAM) and BW/R_S or BS/R_S equal to:

$$\eta = 1 + \alpha = 1.25$$

BW/R_S or BS/R_S ratios which are different from $1 + \alpha$ may be adopted for different service requirements. The adoption of figures significantly lower than $1 + \alpha$ (e.g. $\eta = 1.21$ associated with $\alpha = 0.35$), in order to improve the spectrum exploitation, should be carefully studied on a case-by-case basis, since severe performance degradations may arise due to bandwidth limitations and/or adjacent channel interference, especially with 8PSK and 16QAM modulations and high coding rates (e.g. 5/6 or 7/8).

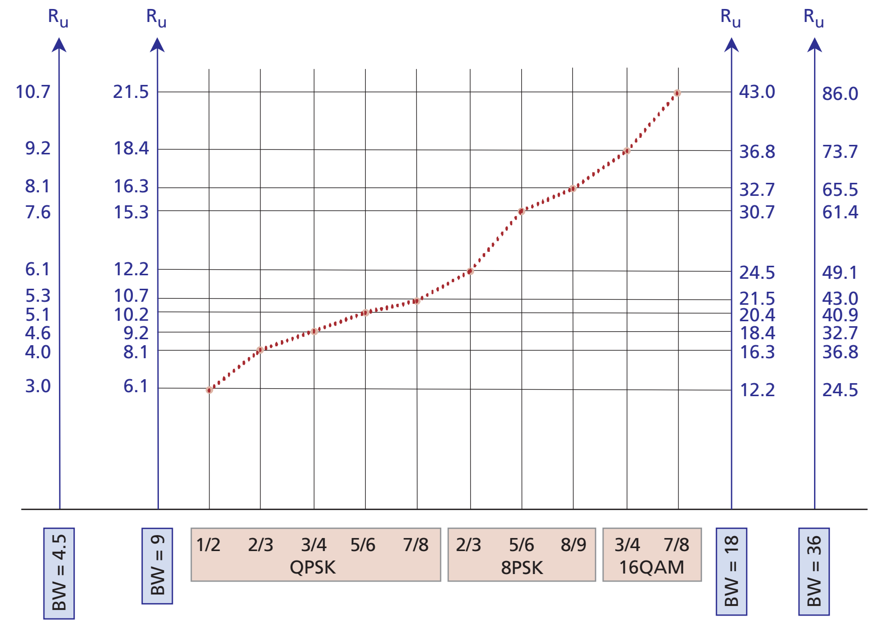

Figure 4
Bit-rate capacity vs. available bandwidth.

Table 2 considers possible examples of the system used in the single-carrier-per-transponder configuration. Different modulation and inner code rates are given with the relevant bit-rates. According to typical practical applications, a BW/R_S ratio equal to 1.31 is considered, offering a slightly better spectrum efficiency than the examples of *Fig. 4* for the same modulation/coding schemes. The 36 MHz transponder bandwidth considered here is wide enough to allow high-quality 422P@ML SCPC transmissions, as well as MP@ML and 422P@ML MCPC transmissions.

Quasi-constant envelope modulations, such as QPSK and 8PSK, are power efficient in a single-carrier-per-transponder configuration, since they can operate on transponders driven near to saturation. Conversely, 16QAM is not power efficient since it can only operate on quasi-linear transponders, i.e. with large OBOs. (In the Appendix to this article, *Fig. A7* shows the E_b/N_0 degradation versus the transponder

input back-off for three modulation and channel coding schemes in a single-carrier-per-transponder configuration.) The use of the narrow roll-off $\alpha = 0.25$ with 8PSK can produce a larger non-linear degradation in a satellite system.

Analogously, *Table 3* considers possible examples of the system used in the multi-carrier FDM configuration and in SCPC mode. Different modulation / coding modes are given with the relevant bit-rates. The E_b / N_0 figures refer to the IF loop specification for QEF operation. The overall linear, non-linear and interference degradations of the satellite should be evaluated on a case-by-case basis; typical figures are of the order of 0.5 to 1.5 dB.

Link budget evaluations have been carried out to estimate the earthstation characteristics required to achieve a suitable service continuity target (i.e. 99.9% or 99.6% of the average year) in Italy, on a typical Ku-band satellite with Europe-wide up-link and down-link coverage. Two Italian *up- link* locations were chosen to represent (i) a typical case (Palermo, ITU climatic zone K) and (ii) a worst case (Turin, ITU climatic zone L); the chosen reception location was Rome (climatic zone K).

To allow a fair comparison of the results, the link budgets were optimized at each location, although it is clear that operation in Italy would require the adoption of a uniform set of transmission parameters, such as the satellite transponder gain setting. For DSNG applications, the up-link antenna diameters were minimized, while neglecting the possibility of receiving the transmitted TV signal by the DSNG terminal. For contribution links connecting fixed stations, the same antenna diameters were adopted at the transmitting and receiving sites, in order to allow a bi-directional exchange of programme material.

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The following link characteristics were adopted:

Up-link Terminals

See *Table 4*.

Parameter	Value
Locations	Turin (zone L) Palermo (zone K)
Frequency (GHz)	14.29
Antenna efficiency (%)	60
Coupling loss (dB)	0.3
Pointing loss (dB)	0.3
OBO (dB): **– QPSK / 8PSK** **– 16QAM**	 2 6

Table 4
Characteristics of the up-link terminal.

Up-link propagation

The atmospheric loss and rain attenuation were:

- ➩ 0.2 + 5.6 dB (Turin) and 0.1 + 3.9 dB (Palermo) for 99.9% of an average year (ay)
- ➩ 0.2 + 2.9 dB (Turin) and 0.1 + 2.0 dB (Palermo) for 99.6% ay.

Satellite

The G / T (dB / °K) were:

- ➩ 4.3 (Turin);
- ➩ 3.6 (Palermo).

The IPFD for saturation (from the –0.5 dB / °K contour) was:

- ➩ variable (–80 dBW / m^2 nominal gain setting).

The transmitted EIRP at saturation was:

- ➩ 46.5 dBW (to Rome);

Down-link propagation

The atmospheric loss and rain attenuation at the Rome site were:

- ➩ 0.1 + 2.4 dB for 99.9% ay;
- ➩ 0.1 + 1.2 dB for 99.6% ay.

Receiving Station

See *Table 5*.

Parameter	Value
Location	Rome (zone K)
Frequency (GHz)	11.99
Antenna efficiency (%)	60
Coupling loss (dB)	0.5
Pointing loss (dB)	0.5
LNB noise figure (dB)	1.1

Table 5
Characteristics of the receiving station.

Table 6
Example use of the system for DSNG and fixed contribution applications: N digital signals in FDMA in a 36 MHz transponder.

	Signals			Up-link terminal						Satellite			Rx station
	Useful bit-rate (Mbit/s)	**Modul. & coding**	**N**	**Target service availa-bility ¹ (%)**	**Type**	**ITU clim. zone**	**HPA power ² (W)**	**Anten. diame-ter (m)**	**EIRP ³ dBW**	**IPFD ⁴ (dBW/m2)**	**IBO ⁵ per carrier (dB)**	**OBO ⁵ total (dB)**	**Antenna diameter (m)**
1	**8.448**	QPSK 3/4	4	99.9	DSNG flyaway	L K	110 70	0.9	58.5 56.5	-84 -87	15.7 15.2	4.2 3.9	3
2	**21.50**	QPSK 7/8	2	99.9	DSNG vehicle	L K	100 70	1.5	62.5 61.0	-82 -86	13.7 11.8	3.7 2.7	4
3	**20.48**	8PSK 5/6	3	99.9	DSNG vehicle	L K	230 90	2.4	70.2 66.1	-70 -74	18.0 18.6	6.6 7.1	6
4	**15.357**	8PSK 5/6	4	99.9	DSNG vehicle	L K	300 75	2.4	71.4 65.3	-68 -74	18.9 19.4	6.8 7.2	6
5	**18.43**	16QAM 3/4	4	99.9	Fixed contrib.	L K	250 60	7 6	75.9 68.3	-62 -71	20.4 19.4	8.0 7.3	7 6
6	**21.50**	16QAM 7/8	4	99.6	Fixed contrib.	L K	60 70	8 7	70.8 70.3	-67 -68	20.4 20.4	8.1 8.1	8 7

1) Percentage of average year; up-link fading.
2) At saturation.
3) At OBO = 2 dB (QPSK and 8PSK), OBO = 6 dB (16QAM).
4) IPFD at saturation for up-links on the –0.5 dB/°K contour (the nominal gain setting is –80 dBW/m²).
5) Nominal in clear sky.

The link analysis method was based on the figures of *Table 1* (IF loop performance) and on computer simulations to estimate the noise margin losses due to the non-linearity, the input/output signal power levels and the intermodulation interference (C/I) between the signals, following the simplified analysis method described in the Appendix [10][11]. An additional link margin of 1 dB was introduced, in order to cope with possible inaccuracies in the simplified analysis method. The link budgets were balanced to achieve the target service continuity (99.9% or 99.6% of the average year) under up-link fading; subsequently the availability of positive margins were verified under down-link fading (for the same service continuity target).

Table 6 shows the results of this analysis for a 36 MHz transponder.

From the examples of *Table 6*, the following considerations may be drawn. For DSNG applications, four QPSK-3/4 signals at 8 Mbit/s may be placed in a 36 MHz transponder (9 MHz frequency slots, see the first row in *Table 4*). In this configuration, very small *fly-away* up-link terminals may be used, with EIRPs in the range 56 - 59 dBW, and 3 m receiving antennas. When higher picture quality is needed (e.g. when using 422P@ML at bit-rates of 21.5 Mbit/s) while at the same time keeping the DSNG up-link antenna small (1.5 m), the satellite bandwidth exploitation has to be reduced from four to two FDM signals (see row 2 in *Table 6*). This configuration requires a larger receiving antenna (4 m). Using 8PSK 5/6 signals (see rows 3 and 4 in *Table 6*), three to four carriers may share the satellite transponder, offering bit-rates of about 20 Mbit/s and 15 Mbit/s, respectively. These configurations require large vehicle-mounted DSNG up-link stations (2.4 m antennas) and large receiving antennas (6 m). Significantly better results, in terms of the requested antenna diameters, may be obtained using satellites with smaller up-link coverage (e.g. national instead of pan-European), since the higher G/T of these satellites directly improves the up-link performance.

For fixed contribution links, high bit-rates (422P@ML video) and high spectrum efficiencies are often required. In the examples of *Table 6* (rows 5 and 6), four 16QAM signals at 18.4 or at 21.5 Mbit/s are allocated in 9 MHz frequency slots, using large transmitting and receiving stations (6 to 8 m antennas). At 21.5 Mbit/s, due to the high C/N+I requirements of 16QAM-7/8, a slightly reduced service availability is accepted in order to keep the antenna diameters at a reasonable level.

It should be noted that in typical operational environments, the optimization of the transponder gain setting (see "IPFD at saturation" in *Table 6*) is limited to about ±3 dB with respect to the *nominal* gain setting, in order to keep the up-link power levels balanced in cross-polar transponders and to avoid severe interference problems on the up-link. Nevertheless, in the given

examples, a significantly wider adaptation has been allowed (in the range +7 to –18 dB), requiring careful interference handling by the satellite operator. This is necessary with high-level modulations, demanding both high C/N+I ratios on the up-link and good transponder linearity.

8. Conclusions

The DVB-DSNG system described here offers significant advantages in terms of picture quality (MPEG-2 coding with 4:2:0 and 4:2:2 image formats), modulation/coding flexibility and rapid link set-up for DSNG applications.

Thanks to its flexibility, it allows the required trade-offs between ruggedness against noise/interference and spectrum efficiency to be achieved. For example, on a typical Pan-European satellite, one to four digital TV signals may be allocated within a 36 MHz transponder, in frequency division multiplex (FDM) format. The resulting link budgets indicate that, when using QPSK modulation, DSNG services at 8 Mbit/s may be established with small "fly-away" terminals using 0.9 m antennas.

When higher picture quality is required (e.g. from 15 to 21 Mbit/s), DSNG services may be established by vehicle-mounted terminals (1.5 - 2.4 m antennas) if QPSK or 8PSK modulation is used. In the case of fixed-contribution links at high bit-rates (e.g. 18 to 21 Mbit/s), 16QAM modulation may be chosen to increase the space segment exploitation, but with a requirement for larger transmitting/receiving antennas (e.g. from 6 to 8 m).

The new DVB-DSNG standard represents a significant step forward in Digital Satellite News Gathering and for fixed contribution links by satellite.

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Appendix: Simplified analysis method

A simplified analysis method has been developed [10] in order to allow a first estimation of the system performance under different operating conditions (e.g. up-link EIRP, nominal TWTA input back-off, noise power density levels etc.), without the need to perform complete computer simulations. A nominal channel spacing respectively of 18 MHz in the case of two carriers per transponder, 12 MHz for three carriers and 9 MHz for four carriers is adopted. The analysis method is focused on signal *b* (see *Fig. A1*), which means the central signal in the three-carriers-per-transponder configuration, and the second signal *b* in the two- and four-carriers-per-transponder configuration.

Fig. A2 gives the AM/AM and AM/PM characteristics of the TWTA, while *Fig. A3* shows the frequency responses of the input multiplexer (IMUX) and the output multiplexer (OMUX) which were adopted in the simulations. The total usable transponder bandwidth was of the order of 36 MHz (at the –3 dB points) and the total group delay at the edge of this band was around 50 - 60 ns.

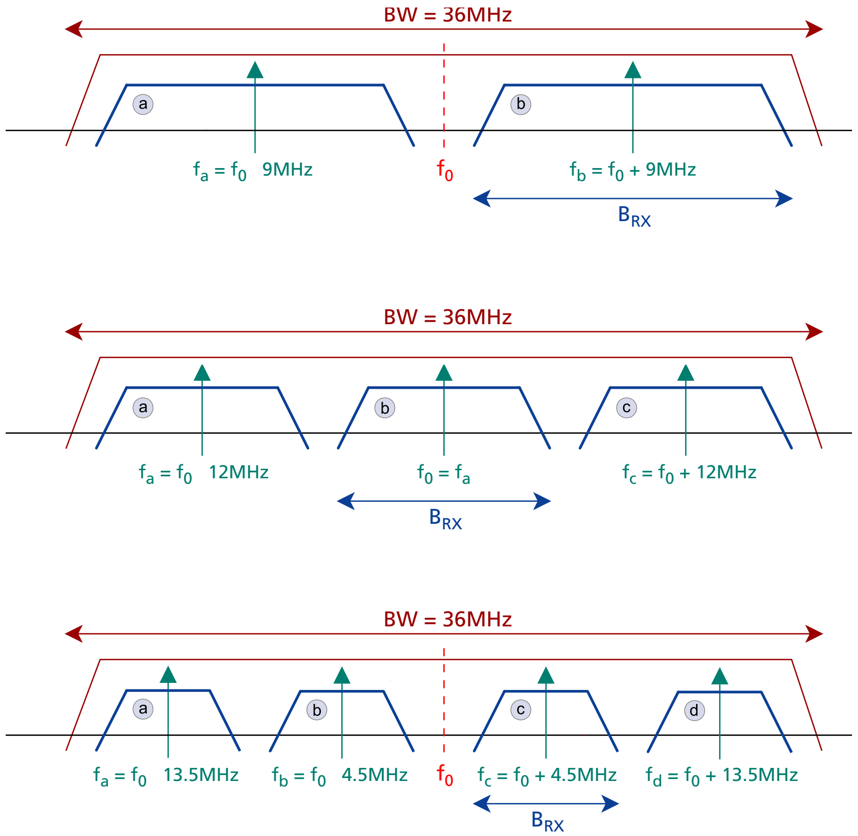


Figure A1
FDM configurations in a 36MHz transponder in the case of two, three and four carriers.

When multiple signals are transmitted in an FDM within a single transponder, the generated TWTA output power is split between the signals according to (i) their input level and (ii) the TWTA AM / AM non-linear characteristic. *Fig. A4* gives OBO_b (the output back-off of signal *b* with respect to the transponder output saturation power) as a function of IBO_b (the input back-off of signal *b* with respect to the transponder input saturation power), for different values of the input back-off of the interfering signals $IBO_{a,c,d}$. The results, obtained by computer simulations, refer to a configuration with four signals in the transponder, where interfering signals were modulated using QPSK, but signal *b* was unmodulated (*Fig. A5*). The power of signal *b* was measured after a narrow-band filter (200 kHz) which suppressed the interference from the other signals. In a first approximation, the cases where N = 2 and N = 3 signals per transponder can be derived from *Fig. A4* through the formula:

$$OBO_b(N) = OBO_b(4) + 10 \log_{10}(N/4)$$

8PSK and 16QAM modulations give approximately the same results and thus *Fig. A4* may be used also for these modulations.

In the simplified analysis method, the following sources of degradation are considered:

a) *Gaussian noise:*

Neglecting the noise compression on the satellite TWTA, the assumption is made that up-link and down-link noise of equal power have the same effect on the system BER, and their powers can be freely added.

b) *Interference (intermodulation) from adjacent signals in FDM:*

Assuming that the channel spacing is larger than the total signal bandwidth including the roll-off, the mutual interference between the signals on a quasi-linear up-link is negligible.

On the down-link, the equivalence and additivity of noise and interference is assumed. In other words an interfering signal of power I_{BRX} (in the receiver bandwidth) is assumed to produce the same BER as a Gaussian noise of equal power (i.e. $N_{BRX} = I_{BRX}$). This approximation neglects the fact that the envelope of the intermodulation signals is not Gaussian and that it is correlated with the signal itself. It should be noted that the noise and interference contributions must be evaluated and added after the demodulator receiving filter (with noise bandwidth B_{RX} equal to the symbol rate R_s). A practical problem is how to measure the interference power (I_{BRX}) in the demodulator filter without removing the useful signal or modifying the TWTA working point.

In the computer simulations, while the interfering signals were modulated, the wanted signal was un-modulated (see *Fig. A5*). The interference power I_{BRX} has been measured by filtering the received signal through a notch filter, centred at the frequency of the unmodulated useful signal, with rejection bandwidth 200 kHz. The parameter C_b corresponds to the TWTA saturation power attenuated by OBO_b. *Fig. A6*, obtained by computer simulations, shows – for the different FDM configurations and modulation schemes – the $(C/I)_b$ curves for the wanted signal *b* versus IBO_b, for different values of $IBO_{a,c,d}$ of the other signals.

c) *Inter-symbol interference:*

The inter-symbol interference (ISI) produced by the TWTA non-linearity depends on the working point (IBO) which, in turn, is determined not only by the signal itself, but also by the other signals that are multiplexed in the transponder. In the simplified analysis described here, it is assumed that the ISI degradation in the FDM configuration is the same as for the single-signal configuration for the same total IBO (the power sum of the input signals). For a single carrier, the noise margin loss Δ_{ISI} with respect to the AWGN channel tends to 0 dB for high back-offs (quasi-linear TWTA) while, approaching the TWTA saturation, the values depend on the modulation and coding rate.

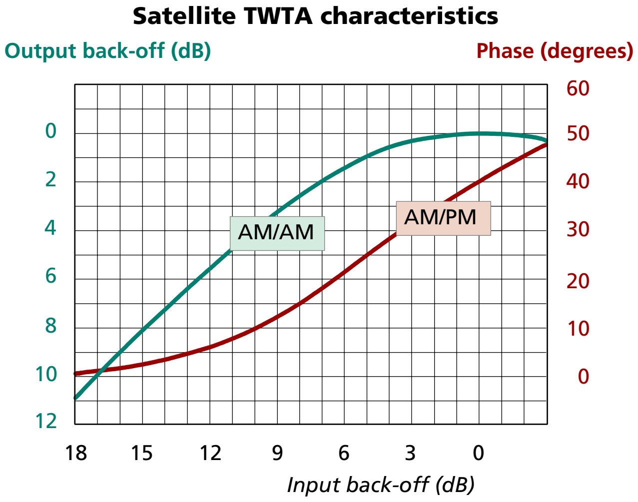


Figure A2
Simulated TWTA AM/AM and AM/PM characteristics.

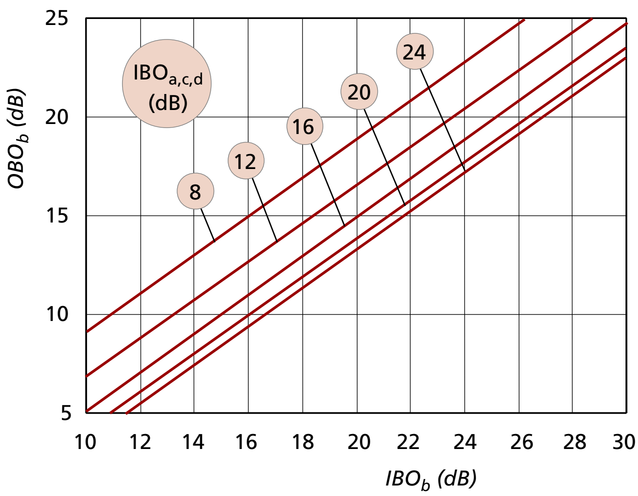


Figure A4
OBO$_b$ vs. IBO$_b$ for different values of IBO$_{a,c,d}$ in the case of four signals per transponder (simulation results).

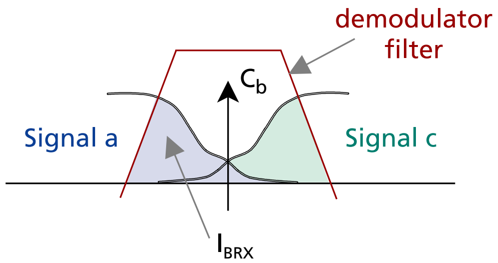
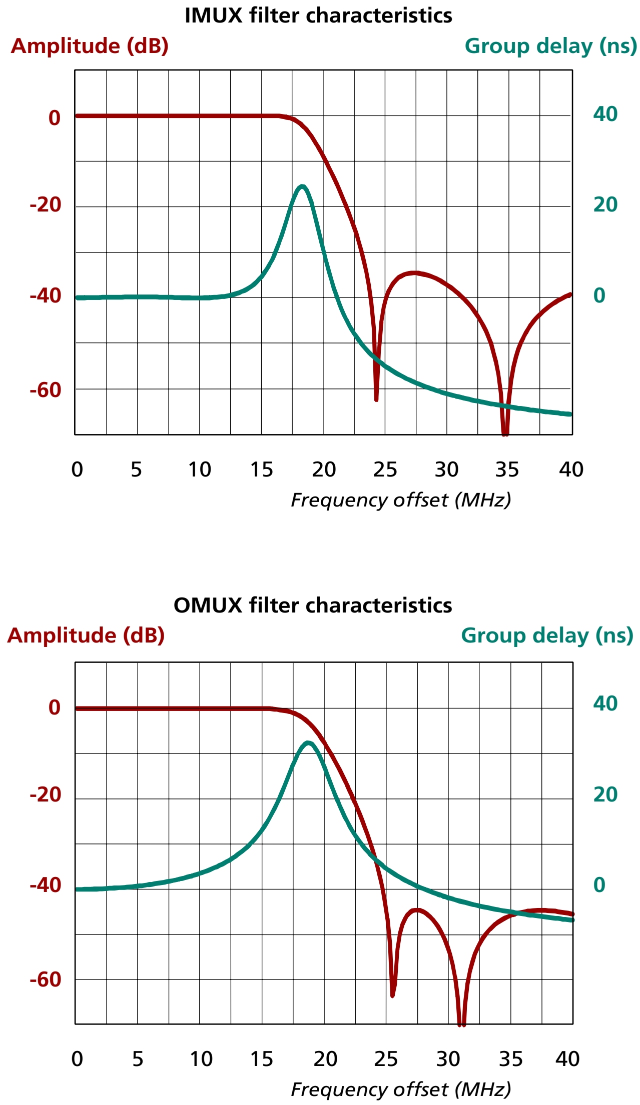


Figure A5
Measurement of the interfering power I_{BRX} in the demodulator filter.

Figure A3
Simulated IMUX and OMUX amplitude and group delay characteristics.



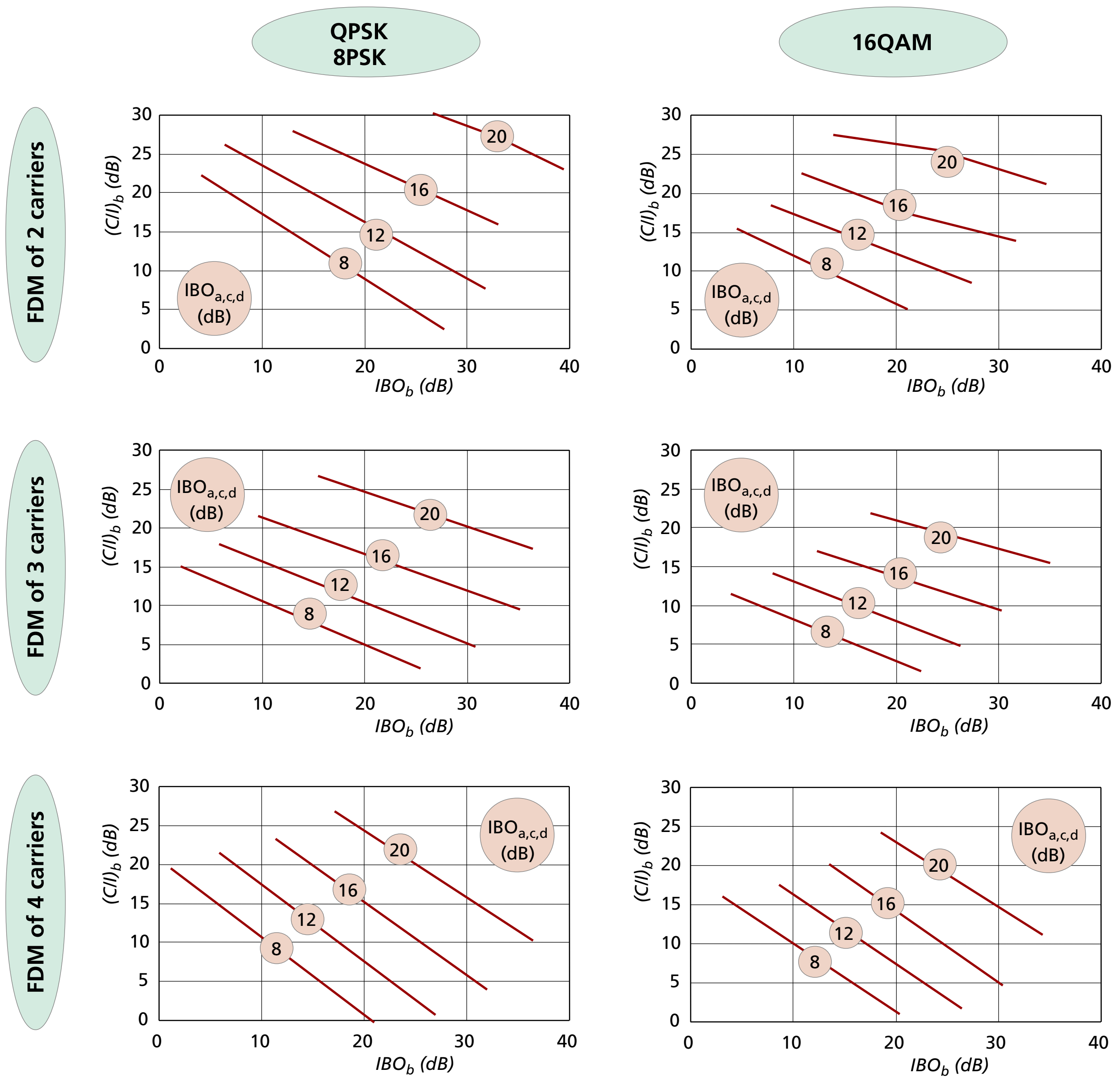


Figure A6
$(C/I)_b$ curves vs. IBO_b for different values of $IBO_{a,c,d}$ (simulation results).

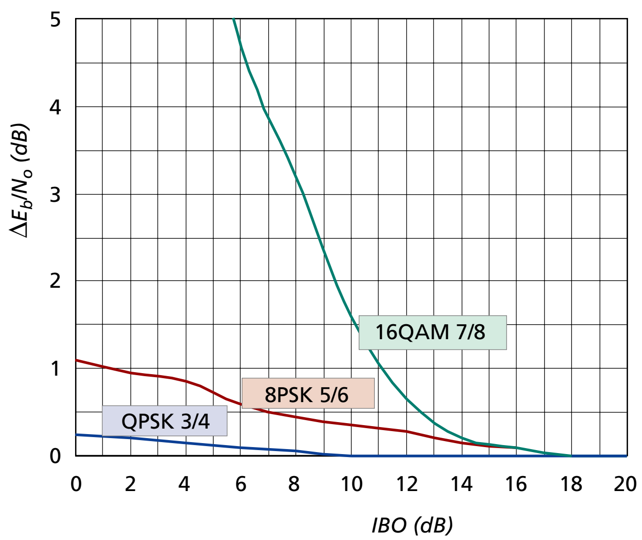


Figure A7
Noise margin loss $\Delta E_b/N_0$ vs. TWTA IBO.

Step	What to evaluate:	Formulae, constants & figures to use
Evaluation of the required C/(N + I) in the noise bandwidth B_{RX}		
1	Required E_b/N_0 on AWGN at BER = 2×10^{-4}	From *Table 1* (IF loop) + 1 dB margin, to allow for possible inaccurancies of the simplified analysis method
2	IBO_{tot}	Add TWTA input powers, normalized with respect to the saturation point
3	ISI noise margin loss on TWTA at IBO_{tot}	from *Fig. A7*
4	Required E_b/N_0 by satellite at IBO_{tot}	Combine the results from 1 and 3
5	Required C/(N + I) by satellite in B_{RX}	C/(N + I) (required) = $E_b/N_0 + 10 \text{ Log}(R_u/B_{RX})$
Evaluation of the available C/(N + I) in the noise bandwidth B_{RX}		
6	OBO_b versus IBO_b , $IBO_{a,c,d}$	from *Fig. A4*
7	Available C/N_u and C/N_d in B_{RX}, at IBO_b and OBO_b	$E_b/N_{0,u} = EIRP_u - A_u - L_u - PL_u - AL_u + G/T_s - k - R_u$ $E_b/N_{0,d} = EIRP_{sat} - OBO_b - A_d - L_d - PL_d - AL_d + G/T_{RX} - k - R_u$ $C/N_u = E_b/N_{0,u} + 10 \text{ Log}(R_u/B_{RX})$ $C/N_d = E_b/N_{0,d} + 10 \text{ Log}(R_u/B_{RX})$ where: k = Boltzmann constant A = rain attenuation L = free space loss PL = pointing loss AL = atmospheric loss The sub-script "u" refers to the up-link and "d" to the down-link
8	$C/I_d(b)$ (intermodulation) in B_{RX} at $IBO_{a,b,c}$	from *Fig. A6*
9	Available $C/(N_u + N_d + I_u + I_d)$	Combine results from 7 and 8
The service quality is guaranteed if C/(N+I) (available) > C/(N+I) (required)		

Table A1
Simpified analysis method.

In *Fig. A7* the noise margin loss is given with respect to the performance on an AWGN channel for the three different modulation schemes of the DVB-DSNG system, for one representative coding rate. In link-budget computations, since the variation of Δ_{ISI} with the coding rate is very low (about 0.1 dB at the typical quasi-linear TWTA operating points in FDM configurations), the curves of *Fig. A7* have been adopted for any coding rate.

d) Cross-polar co-channel digital interference:

It is assumed that cross-polar co-channel digital interference has similar effects on the system BER as Gaussian noise with equal power in the receiving filter. When the cross-polar transponder carries the same signal configuration as the wanted transponder, the interference power (I) is equal to the wanted power (C) attenuated by the cross-polar antenna discrimination (XPD): i.e. C/I = XPD.

The simplified analysis method described in this Appendix adds all the interference, intermodulation and noise power contributions (measured within the receiver filter bandwidth) in order to evaluate the *available* C/(N+I) ratio on the satellite links, and compares it with the C/(N+I) ratio *required* by the modulation system to deliver a target BER of 2×10^{-4}. The service continuity and quality is assured when C/(N+I) (available) > C/(N+I) (required). The detailed computation procedure is summarized in *Table A1*.

EBU interoperability tests
– DSNG equipment based on MPEG-2

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In October 1997, the EBU carried out tests to verify the interoperability of MPEG-2 MP@ML satellite news-gathering equipment from various manufacturers – under operational conditions via an 8.448 Mbit/s satellite channel.

A report on these tests was published by the EBU in July 1998 and this article is based on that report.

Original language: English
Manuscript received: 10/7/98.

1. Introduction

Intelsat, in collaboration with ISOG, carried out a series of tests on DSNG equipment in 1996/7 at their technical laboratories in Washington DC. A complete description of these tests has been published by Intelsat [1] and only an extract of the objectives and results is reproduced in this article.

The last round of these tests was designed to achieve "plug and play" operation with both the NTSC and PAL standards, for several different parameter sets. This was a significant extension of previous tests in the series which did not include PAL, did not address "plug and play", and were only conducted for one parameter set.

In testing "plug and play", the only adjustments allowed to be made to the equipment were those that would be available to the typical user during first-time configuration of the equipment. The aim was to document the interoperability of equipment operating within the MPEG-2 and DVB framework, but not to assess compliance with any standard.

The results for interoperation of the equipment are given in Attachment 2 to the Intelsat report [1]. Each page of this attachment consists of a matrix of connections from one encoder system to multiple decoder systems under different operating parameters, such as the symbol rate, FEC code rate and the video format. The results, in general, are very encouraging and demonstrate which coders and IRDs will "plug and play" at data-rates which might be appropriate for various applications.

The measurements performed through the Intelsat VII simulator showed no degradation of performance for the parameter sets chosen (e.g. 6 Msymbol/s, FEC rate ¾ , PAL). The cases where interoperation was not achieved were due to limitations of the available hardware or software in supporting a particular parameter set.

When the results of these tests were reported at the ISOG meeting in Washington (June 97), there was some discussion about possible further testing. The general feeling was that the Intelsat series of tests had indeed been conclusive and that another round of similar tests would bring no further evidence, but would just impose an additional burden on the participating manufacturers and Intelsat.

However, ISOG did identify two possible ways of extending these tests:

- ⇨ by using new technical standards such as MPEG-2 4:2:2P@ML and/or using 8-PSK modulation techniques.
- ⇨ by trying to repeat the tests (made in the laboratory in the presence of manufacturers' engineers) in a real environment with a satellite link, SNG stations and many receiving points.

In response to this "suggestion", the EBU volunteered to make its *Eurovision* network available in order to repeat the Intelsat tests under operational conditions, using a real satellite with links to and from various earthstations and SNG stations.

2. The EBU tests

During the summer of 1997, an invitation to participate in these EBU tests

Date	Time: AM/PM	Tx Source	Tx earthstation	Tx equipment	Eutelsat Approval No.
6 Oct.	PM	EBU/ GNVE	PTT-CH 1.8m	NDS-DSNG	SUI-001
7 Oct	AM/PM	ZDF/ MANZ	SweDish 0.9m	NDS-DSNG	D-100
8 Oct	AM/PM	ARD/ FFTM	Euroradio 4.2m	Thomson	D-009
9 Oct	AM/PM	BRT/ BRUX	Euroradio 3.7m	Tiernan T-E 3	BEL-BRU-012
10 Oct	AM	BBC/ LNDN	BBC 9m	Wegener DVT 2000	UKI-TVC-002
10 Oct	PM	BBC/ LNDN	BBC 9m	Nextlevel Systems SE-3200	UKI-TVC-002
13 Oct	AM	NBC/ LNDN	Advent Mantis 1.9m	NDS-DSNG	UKI-193
13 Oct	PM	NBC/ LNDN	Advent Mantis 1.9m	Wegener DVT-2000	UKI-193
14 Oct	AM/PM	Tadiran/ ISR Scopus	PTT-ISR 2.4m	T-S E-110	ISR-1
15 Oct	AM	SVT/STOK	SweDish 0.9m	Tiernan T-E 3	SWE-11
16 Oct	AM	NTL/ LNDN	Steerable 5.6m	SA PowerVu	UK-WIN-001
17 Oct.	AM/PM	YLE/HLKI	SweDish 0.9m	DMV 3000	FIN-Temp-002

Table 1
EBU test programme over several days.

was sent to all EBU members and all the manufacturers who had been involved in the Intelsat tests. It was clearly stated that these new tests were open to anyone willing to participate and they would also be conducted on a voluntary basis.

The tests were carried out over several days in October 1997, using an 8.448 Mbit/s satellite channel. Each day, one SNG earthstation transmitted MPEG-2 MP@ML signals using an encoder/modulator combination from a different manufacturer, as shown in *Table 1*.

2.1. Transmitting equipment

The following manufacturers made their equipment available for transmission purposes:

- Tiernan;
- NDS (ex-DMV);
- Thomson;
- Wegener;
- Nextlevel;
- Tadiran-Scopus;
- Scientific Atlanta.

In addition, IRDs from various manufacturers were utilized for reception, and MPEG-2 Transport Stream Analyzers from the following manufacturers were used to monitor the signals:

- Adherent Systems Ltd at BBC/LNDN;
- Hewlett-Packard at BRT/BRUX;
- Snell & Wilcox at EBU/GNVE (9 and 10 October only).

2.2. Receiving stations

The following locations on the *Eurovision* network were equipped with at least one IRD in order to receive the signals:

- Germany – Frankfurt (ARD/FFM);
- Belgium – Brussels (BRT/BRUX);
- Finland – Helsinki (YLE/HLKI);
- Sweden – Stockholm (SVT/STOK);
- Switzerland – Geneva (EBU/GNVE);
- UK – London (BBC/LNDN).

Furthermore, the signals were also received in Israel by Tadiran-Scopus in collaboration with the national carrier (PTT/ISR) and also in Ireland by Amstrong.

All these stations sent reports to the EBU in Geneva and these are summarized in the Appendix to this article.

The signals received by the six EBU *Eurovision* stations were also re-injected in analogue format via a *Eurovision* satellite channel when available, and these analogue signals were monitored by EBU/GNVE.

2.3. Technical parameters

The transmission parameters were as follows:

Composite bit-rate	8.448 Mbit/s (including Reed-Solomon)
Modulation rate	5.632 Msymb/s
Modulation	QPSK
FEC	¾
Audio coding	MPEG Layer II, 256 kbit/s

The EUT-P channel [1] was made available between 07.00 - 09.00 and 12.00 - 14.00 GMT from 6 - 10 October and from 13 - 17 October 1997.

Up-link frequency (X polarization)	14 341 MHz
Down-link frequency (Y polarization)	11 041 MHz

The up-link EIRP was set to give 16 dB IBO at the satellite input, (i.e. 67 dBW at the 0.5 dB/°K contour). In the case of the transportable earthstations, an

1. The EBU-leased transponder 25 of the Eutelsat II F-4 M satellite at 7° East.

IFLU was carried out with Eutelsat / CSC in Paris at the start of each day's transmission as required.

2.4. *Test procedures*

Ten 30-minute Betacam test cassettes were prepared at EBU / GNVE and sent to the various originating sources (see *Table 1*). These cassettes consisted of the EBU moving-picture test sequences (nos. 30 to 53) plus five minutes of a "President Clinton – State of the Union" speech, for lip-sync verification purposes.

The test sequences had 1 kHz tone recorded at EBU "Test Level" (i.e. 9 dB below "peak permitted level") on the left audio track, and 500 Hz tone recorded at EBU "Test Level" on the right track.

Abbreviations

DSNG	Digital satellite news gathering
DVB	Digital Video Broadcasting
IEC	International Electrotechnical Commission
EIRP	Effective isotropic radiated power
FEC	Forward error correction
IBO	Input back-off
IFLU	Initial full line-up
IRD	Integrated receiver/ decoder
ISO	International Organization for Standardization
ISOG	Inter-Union Satellite Operations Group
MPEG	(ISO/IEC) Moving Picture Experts Group
QPSK	Quadrature (quaternary) phase-shift keying
SDI	Serial digital interface
SNG	Satellite news gathering

Each source sent this material at least once. They then sent additional material, usually of sporting events, from local sources.

The input signal to the encoder was PAL in most cases, but an SDI input signal with a true component source was also utilized when available. A 525-line test pattern, from an NTSC source, was transmitted on 13 October from NBC / LNDN. All IRDs automatically decoded the 525-line signal without problems.

To ensure that all the receiving points were aware of the type of video signal being fed to the encoder, the type of encoder / modulator being used and the source location information was inlaid on a test pattern at the start of each transmission.

3. Conclusions

Since the signals were received at various locations, it is inevitable that the attention given to monitoring the video and audio was more assiduous at some locations than at other locations. Therefore, the fact that some receiving sites have noted more incidents of degradation than others does not necessarily mean that the encoder-decoder combinations concerned are less compatible than other combinations.

Interoperability was demonstrated for all tested combinations of encoder / modulator and IRD respectively. Even in cases where one IRD was not correctly receiving the signal at one location, there was always another location where a similar IRD was able to receive the signal satisfactorily.

The most common problem encountered was spectrum inversion, requiring the demodulator to be selected to "Inverted Spectrum" before a signal could be received. However, some IRDs will automatically correct for spectrum inversion.

Another common problem encountered was audio left / right inversion.

Audio levels delivered by the IRDs were up to 12 dB too high and 12 dB too low in some cases.

The video sequences were correctly encoded and decoded for all combinations of encoder / decoder most of the time, although macroblocks were observed in some cases, and a few instances of picture-freezing were also observed. These symptoms suggest that some decoders may have inadequate buffer capacity, where the encoder and decoder buffers do not match.

When one particular encoder was fed from an SDI component source, there were occasions – following scene changes – when some decoders produced visible and audible decoding errors. However, the decoder from the same manufacturer as the encoder had no problem decoding these critical sequences. True component source signals have higher entropy than equivalent composite signals, mainly due to the limited chrominance bandwidth of composite signals. Scene changes represent entropy peaks, so it is not surprising that limitations of encoder-decoder interoperability are noticeable in these circumstances.

The tests demonstrated one of the key advantages of MPEG-2 systems, i.e. the ability of MPEG-2 decoders to adjust automatically to the coding parameters. For example, the actual bit-rate used for video encoding varied between 5.7 Mbit / s and 7.14 Mbit / s, but the decoders adapted to each bit-rate automatically.

On the other hand, differences in the way the composite bit-rate and / or the symbol rate are specified led in some cases to problems in correctly setting up the encoder / modulators and the IRDs.

All the encoder-decoder combinations tested by the EBU have demonstrated correct interoperability. However, as the manufacturers concerned have been making improvemants to encoder and decoder software over a period of time, to achieve this goal, users should make sure that they have up-to-date software versions in order to ensure optimum interoperability.

Bibliography

[1] Intelsat/ISOG: **Digital Video Transmission Equipment Interoperation Tests**
Tests Report, March 1997.
http://www.intelsat.int/index.htm

Appendix:
Reports sent in to the EBU from the receiving stations

6 October 1997 – PM

Source: EBU/GNVE
Encoder/Modulator: NDS-DSNG
Tx earthstation: PTT/CH 1.9m (SUI-001)

RX point	IRD	Video	Audio	Lip-Sync	Remarks
BBC/LNDN	Wegener DVR2000 Next Level	Good Good	Good Good	OK OK	
BRT/BRUX	NDS	Good	Good	OK	Audio 2 dB high
SVT/STOK	Tiernan TDR-7	----------	----------	------------	Misunderstanding about composite bit-rate
YLE/HLKI	Tandberg TT1200	Good	Good	OK	
ARD/FFTM	--------------------	-----------	----------	------------	Equipment had not yet arrived at ARD/FFTM
EBU/GNVE	NDS-3000	Good	Good	OK	
Tadiran/ISR	Scopus IRD-250	Good	Good	OK	
Armstrong/IRL	Scopus IRD-250 STS	Good Good	Good Good	OK OK	

7 October 1997 – AM/PM

Source: ZDF/MANZ
Encoder/Modulator: NDS-DSNG
Tx earthstation: SWE-DISH 0.9 m

RX point	IRD	Video	Audio	Lip-Sync	Remarks
BBC/LNDN	Wegener DVR 2000 Next Level	Good Good	Good Good	OK OK	
BRT/BRUX	Tiernan TDR-7	Fair	Good	OK	Macroblocks observed during one critical SDI sequence.
SVT/STOK	Tiernan TDR-7	Fair	Good	OK	Some blocking observed following scene cuts. Audio 6 dB high.
YLE/HLKI	Tandberg TT1200	Good	Good	OK	
ARD/FFTM	NDS-DSNG	Good	Good	OK	Equipment ready PM only.
EBU/GNVE	NDS-3000	Good	Good	OK	
Tadiran/ISR	Scopus IRD-250	Fair	Good	OK	Some errors observed during SDI sequence.
Armstrong/IRL	Scopus IRD-250 STS	Fair Fair	Good Good	OK OK	Break up on peak whites. Break up on peak whites.

8 October 1997 – AM/PM

Source: ARD/FFTM
Encoder/Modulator: Thomson DBE2110/DBM 3221 (Software version 5.0)
Tx earthstation: Euroradio 4.2 m

RX point	IRD	Video	Audio	Lip-Sync	Remarks
BBC/LNDN	Wegener DVR 2000 Next Level SA PowerVu	Good Good Good	Good Good Good	OK OK OK	
BRT/BRUX	Tiernan TDR-7	Fair	Good	OK	Some macroblocks observed on critical sequences.
SVT/STOK	Tiernan TDR-7	Good	Good	OK	
YLE/HLKI	Tandberg TT1200	Good	Good	OK	
ARD/FFTM	NDS-DSNG	Good	Good	OK	
EBU/GNVE	NDS-3000	Good	Good	OK	Left/Right audio inverted.
Tadiran/ISR	Scopus IRD-250	Fair	Good	OK	Occasional freezing observed on critical sequences.
Armstrong/IRL	Scopus IRD-250 STS	Fair Fair	Good Good	OK OK	Problem with movement. Problem with movement.

9 October 1997 – AM/PM

Source: BRT/BRUX
Encoder/Modulator: Tiernan TE 3
Tx earthstation: Euroradio 3.7 m

RX point	IRD	Video	Audio	Lip-Sync	Remarks
BBC/LNDN	Wegener DVR2000 Next Level SA PowerVu	Good Good Good	Good Good Good	OK OK OK	
BRT/BRUX	Tiernan TDR-7	Good	Good	OK	
SVT/STOK	Tiernan TDR-7	Good	Good	OK	Audio 8 dB high.
YLE/HLKI	Tandberg TT1200	Good	Good	OK	
ARD/FFTM	NDS-DSNG	Good	Good	OK	
EBU/GNVE	NDS-3000	Fair	Good	OK	Spectrum inverted. Blocking during "Diva plus noise".
Tadiran/ISR	Scopus IRD-250	Fair	Good	OK	Blocking during"Diva plus noise".
Armstrong/IRL	Scopus IRD-250 STS	Fair Fair	Good Good	OK OK	Some errors observed. Some errors observed.

10 October 1997 – AM

Source: BBC/LNDN
Encoder/Modulator: Wegener
Tx earthstation: Multipoint Vertex

RX point	IRD	Video	Audio	Lip-Sync	Remarks
BBC/LNDN	Wegener DVR 2000 Next Level SA PowerVu	Good Good Good	Good Good Good	OK OK OK	
BRT/BRUX	Tiernan TDR-7	Fair	Good	OK	Blocking at top of screen.
SVT/STOK	Tiernan TDR-7	Fair	Good	OK	Blocking at top of screen and following scene changes. Audio 9 dB high.
YLE/HLKI	Tandberg TT1200	Good	Good	OK	
ARD/FFTM	NDS-DSNG	----------	---------	------------	Spectrum inverted – ARD not aware.[1]
EBU/GNVE	NDS-3000	Good	Good	OK	Spectrum inverted.
Tadiran/ISR	Scopus IRD-250	----------	----------	------------	Public Holiday in Israel.
Armstrong/IRL	Scopus IRD-250 STS	Good Good	Good Good	OK OK	

1) The EBU/GNVE NDS-DSNG IRD decoded the signal correctly

10 October 1997 – PM

Source: BBC/LNDN
Encoder/Modulator: Next Level Systems SE-3200
Tx earthstation: Multipoint Vertex

RX point	IRD	Video	Audio	Lip-Sync	Remarks
BBC/LNDN	Wegener DVR 2000 Next Level SA PowerVu	Good Good Good	Good Good Good	OK OK OK	
BRT/BRUX	Tiernan TDR-7	Good	Good	OK	
SVT/STOK	Tiernan TDR-7	Good	Good	OK	
YLE/HLKI	Tandberg TT1200	Good	Good	OK	
ARD/FFTM	NDS-DSNG	Good	Good	OK	Spectrum inverted.
EBU/GNVE	NDS-3000	Good	Good	OK	Spectrum inverted. Audio -6 dB and L/R inverted.
Tadiran/ISR	Scopus IRD-250	----------	----------	------------	Public Holiday in Israel.
Armstrong/IRL	Scopus IRD-250 STS	Good Good	Good Good	OK OK	

13 October 1997 – AM

Source: NBC/LNDN
Encoder/Modulator: NDS-DSNG
Tx earthstation: Advent Mantis 1.9 m

RX point	IRD	Video	Audio	Lip-Sync	Remarks
BBC/LNDN	Wegener DVR 2000 Next Level SA PowerVu	Good Good Good	Good Good Good	OK OK OK	
BRT/BRUX	Tiernan TDR-7	Good	Good	OK	
SVT/STOK	Tiernan TDR-7	Good	Good	OK	Audio 5 dB high
YLE/HLKI	Tandberg TT1200	Good	Good	OK	
ARD/FFTM	NDS-DSNG	Good	Good	OK	
EBU/GNVE	NDS-3000	Good	Good	OK	
Tadiran/ISR	Scopus IRD-250	Good	Good	OK	
Armstrong/IRL	Scopus IRD-250 STS	Good Good	Good Good	OK OK	

13 October 1997 – PM

Source: NBC/LNDN
Encoder/Modulator: Wegener DVT-2000 (Version 2.2.5 firmware)
Tx earthstation: Advent Mantis 1.9 m

RX point	IRD	Video	Audio	Lip-Sync	Remarks
BBC/LNDN	Wegener DVR 2000 Next Level SA PowerVu	Good Good Good	Good Good Good	OK OK OK	
BRT/BRUX	Tiernan TDR-7	Good	Good	OK	
SVT/STOK	Tiernan TDR-7	Fair	Fair	OK	Blocking at top of picture and following scene changes. Audio breaks, also noted at same time.
YLE/HLKI	Tandberg TT1200	Fair	Good	OK	Colour flicker and pumping observed.
ARD/FFTM	NDS-DSNG	-----------	----------	-----------	Reception not working. (IRD configuration problem ?) [1]
EBU/GNVE	NDS-3000	Good	Good	OK	
Tadiran/ISR	Scopus IRD-250	Fair	Fair	OK	Two sequences caused decoder failure.
Armstrong/IRL	Scopus IRD-250 STS	Fair Fair	Good Good	OK OK	Lost picture occasionally. Lost picture occasionally.

1) The EBU/GNVE NDS-DSNG IRD decoded the signal correctly.

14 October 1997 – AM

Source: Bezeq/Israel-1
Encoder/Modulator: Tadiran Scopus E-110 (Software 2.53)/EF Data 2020
TX earthstation: PTT-IR 2.8 m

RX point	IRD	Video	Audio	Lip-Sync	Remarks
BBC/LNDN	Wegener DVR 2000 Next Level SA PowerVu	Good Good Good	Good Good Good	OK OK OK	
BRT/BRUX	Tiernan TDR-7	Poor	Good	OK	Intermittent horizontal bar & coloured blocks across top of picture. Many frozen frames and loss of sync.
SVT/STOK	Tiernan TDR-7	Poor	Fair	OK	Blocking at top of picture and following scene changes. Audio breaks also noted at same time.
YLE/HLKI	Tandberg TT1200	Good	Good	OK	
ARD/FFTM	NDS-DSNG	-----------	----------	-----------	Reception not working. (IRD configuration problem?)*
ZDF/MANZ	Tiernan TDR-7 Philips DVS 3824	Fair Good	Good Good	OK OK	Blocking at top of picture
EBU/GNVE	NDS-3000	Good	Good	OK	
Tadiran/ISR	Scopus IRD-250	Good	Good	OK	
Armstrong/IRL	Scopus IRD-250 STS	Good Good	Good Good	OK OK	

15 October 1997 – AM

Source: SVT/STOK
Encoder/Modulator: Tiernan TE-3
TX earthstation: SWE DISH 0.9 m

RX point	IRD	Video	Audio	Lip-Sync	Remarks
BBC/LNDN	Wegener DVR 2000 Next Level SA PowerVu	Good Good Good	Good Good Good	OK OK OK	
BRT/BRUX	----------------------	-----------	----------	------------	No staff available (Belgian holiday)
SVT/STOK	Tiernan TDR-7	----------	---------	-----------	No monitoring whilst transmitting.
YLE/HLKI	Tandberg TT1200	Good	Good	OK	
ARD/FFTM	NDS-DSNG	Good	Good	OK	
EBU/GNVE	NDS-3000	Good	Good	OK	L/R audio inverted.[1]
Tadiran/ISR	Scopus IRD-250	----------	---------	---------	Video unusable, unable to decode properly [2].
Armstrong/IRL	Scopus IRD-250 STS	Good Good	Good Good	OK OK	

1) The EBU/GNVE NDS-DSNG IRD decoded the signal correctly.
2) Maybe due to use of 1.4 m dish for reception. (The Armstrong/IRL Tadiran Scopus IRD-250 decoded the signal correctly).

16 October 1997 – AM

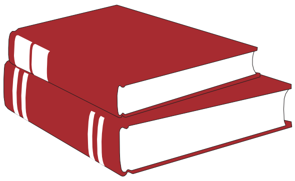
Source: NTL/LNDN
Encoder/Modulator: SA Power Vu
Tx earthstation: Steerable 5.6 m

RX point	IRD	Video	Audio	Lip-Sync	Remarks
BBC/LNDN	Wegener DVR 2000 Next Level SA PowerVu	Good Good Good	Good Good Good	OK OK OK	
BRT/BRUX	Tiernan TDR-7	Good	Good	OK	
SVT/STOK	Tiernan TDR-7	Good	Good	OK	
YLE/HLKI	Tandberg TT1200	Fair	Good	OK	Some colour lag on red areas observed.
ARD/FFTM	NDS-DSNG	Good	Good	OK	
EBU/GNVE	NDS-3000	Good	Good	OK	L/R audio inverted.
Tadiran/ISR	Scopus IRD-250	----------	---------	---------	Holiday in Israel.
Armstrong/IRL	Scopus IRD-250 STS	Good Good	Good Good	OK OK	Lost lock on one occasion. Lost lock on one occasion.

17 October 1997 – AM/PM

Source: YLE/HLKI
Encoder/Modulator: DMV 3000
Tx earthstation: SWE DISH 0.9 m

RX point	IRD	Video	Audio	Lip-Sync	Remarks
BBC/LNDN	Wegener DVR 2000 Next Level SA PowerVu	Good Good Good	Good Good Good	OK OK OK	
BRT/BRUX	Tiernan TDR-7	Good	Good	OK	
SVT/STOK	Tiernan TDR-7	Good	Good	OK	
YLE/HLKI	Tandberg TT1200	Good	Good	OK	
ARD/FFTM	NDS-DSNG	Good	Good	OK	
EBU/GNVE	NDS-3000	Good	Good	OK	One short picture freeze observed.
Tadiran/ISR	Scopus IRD-250	Fair	Good	OK	Some freezing observed.
Armstrong/IRL	Scopus IRD-250 STS	Fair Fair	Fair Fair	OK OK	Occasionally lost lock. Occasionally lost lock.



Bookshelf

For further information about the publications reviewed here, please contact the respective publisher.

Radio Data System

RDS has been one of the EBU's success stories. The first RDS specification was published by the EBU in 1984, after joint developments by several EBU Members in a spirit of mutual co-operation.

Two of the key players in the saga of RDS, Dietmar Kopitz and Bev Marks, have written this comprehensive book about RDS. As well as giving detailed explanations of "how" and "why" RDS works, it traces the history of RDS and its predecessors, such as the ARI system, and looks forward to new applications and services.

An early focus of RDS activities was to assist in the tuning of FM radios, especially in cars. Rather than displaying the frequency of the transmitter (which is of little relevance to the general public), RDS radios display the name of the radio service. Even better, RDS radios automatically re-tune to the best signal carrying the desired programme.

Amongst engineers, the features of RDS are known by a large number of abbreviations, such as PS (Programme Service name), PTY (Programme Type), AF (Alternative Frequency list) and ODA (Open Data Applications). Some RDS radios even have a button marked "TP" (which engineers will recognize as "Traffic Programme") but few consumers will understand what it does! This book explains all the features of RDS – including relatively new features such as DGPS.

The RDS specification has been revised several times since 1984: An early, but very important, change to the original specification was that the ON (Other Networks) method of providing information about other networks was replaced by EON (Enhanced Other Networks). As this book states, the ON mechanism *"just did not work!"*. On the basis that we may learn more from failures than from successes, it is unfortunate that the book provides no further explanation of this problem.

From the early days of RDS, it was clear that RDS could support a variety of additional data services, such as programme-related text services and paging. In recent years, broadcasting of Traffic and Travel Information has become increasingly important, especially for motorists. RDS-TMC was developed to deliver comprehensive information services without consuming large amounts of "air-time" for spoken announcements.

RDS-TMC was also conceived as a method of providing pan-European travel information services since it offered the prospect of a motorist being able to drive throughout Europe whilst hearing relevant traffic messages in his or her own language. This ambitious vision has not yet been realized, partly because voice synthesizers still leave a lot to be desired.

As indicated in this book, an even greater difficulty has been caused by the limited capacity of the RDS data channel. To overcome this problem, RDS-TMC uses a very efficient technique to compress data about traffic events, such as accidents or road closures. The short codes transmitted over the RDS channel are used in conjunction with location databases in each vehicle so that drivers can be given full information about traffic events. In general, the location databases for each country have been, or will be, prepared by national road authorities. It is expected that a modest handling charge will be made to cover the costs of replication and distribution, whether in the form of a smart card or a CD-ROM. However, some owners of national databases may try to recover the full costs of development from motorists, by limiting the use of the database to subscribers to the RDS-TMC services. In addition, there are also concerns about how the databases can be extended and maintained. It is clear that solving technical problems is easy compared with overcoming such institutional problems.

This book is lucid and well-written. I am sure that it will become a valuable reference source for all those involved in RDS. It is strongly recommended.

RDS: The Radio Data System
D. Kopitz and B. Marks
Hardbound volume of 300 pages
Artech House Books, London, 1998
Ref: ISBN 0-89006-744-9. Price £60.00

Phil Laven

Audio test signals

Many years ago, someone remarked that a true engineer would rather listen to "tone" than to any music yet composed. If that is so, then this must surely be their favourite CD. Although one of the most boring CDs ever made, it must also be one of the most useful to engineers working in broadcasting and other related branches of audio engineering.

Made under the supervision of Dr John Emmett, who has led and contributed to many EBU working groups over the years, almost all the recordings on the disk are sine-waves – of almost every imaginable combination of frequencies, levels and phases. These are, of course, test signals for audio equipment and meters of all sorts.

In the informative booklet that accompanies the CD, Dr Emmett leads us through the maze of standards which have been developed over the years by numerous national, international, professional and industry bodies who have tried to bring some order to the apparently simple task of finding and keeping an audio level.

A CD such as this is an ideal tool for providing a source of precision test signals. There is room for nearly 100 tracks so that the correct one can easily be identified and used for the task in hand. The CD does not seek to impose any solutions but, instead, contains examples of all known alignment levels as well as tracks that are suitable for use with all the programme meters in common use.

Briefly, the disk contains stereo tracks with:

- alignment levels and line-up signals;
- system test tracks;
- signals to test VU meters;
- signals to test three different variations of Peak Programme Meters (PPMs);
- signals to test phase meters;
- signals to test loudness meters;
- signals to check peak / RMS levels;
- pink noise;
- male speech.

This disk of course does not set out to replace such disks as the EBU SQAM or PEQS disks. These EBU disks contain music extracts and real sounds and are aimed at the subjective testing of systems and to give examples of quality of production.

The disk under review does however contain some of the same, or similar, signals to the Euroradio Measurements CD, EBU Tech 3270, but the EBU disk is designed for testing circuits as its name suggests.

International Broadcast Standards Tests is available as a DAT and MD as well as a CD.

International Broadcast Standards Test CD
Available on DAT, MD and CD
Canford Audio plc, 1998
http://www.canford.co.uk/

Richard Chalmers

The Tapeless Audio Directory – 7th Edition

The 6th Edition of the Tapeless Audio Directory was reviewed in EBU Technical Review No. 274, Winter 1997.

This edition follows the format of the earlier editions with the new features mentioned below. It contains over 500 entries, covering professional random-access audio systems intended for applications in radio and TV sound. These applications now include location recording, simple cart replacement, comprehensive editing and mixing, live play-out and fully automated systems for radio.

The stated aim of the guide is to help the potential purchaser establish what is on the market. The guide provides, where possible, full information on all important systems gained from questionnaires completed by manufacturers. This information covers target markets, hardware and software specifications, operational features, interfacing, networking and file transfer to external devices, archiving and backup systems. Also included is information on typical configurations and costs, on suppliers in Europe and elsewhere, and on training and support. Manufacturers' plans for future development are included with many of the entries.

In addition, the guide contains explanations of the terminology and offers advice to potential purchasers.

Where possible, this new edition of the guide indicates which hardware is required for any system; it indicates what is supplied as standard, what the user is required to supply, or what is optionally available. For editing systems, the guide now shows which processes are performed in real-time and which need to be rendered. It also now includes information on how systems interface with external devices, and how they can deal with the new longer word sizes and new digital formats.

For broadcast systems, more information is provided on how storage is supplied and organized, and how files are managed for library / database purposes, archiving and file transfer. An increasing number of systems now support BWF (Broadcast Wave Format), developed by the EBU, and the Directory shows which file formats are supported, and whether these can be read directly and / or imported / exported.

This is a useful independent overview of the marketplace and could easily recover its cost by preventing fruitless telephone calls or, indeed, the purchase of inappropriate equipment.

The Tapeless Audio Directory (7th edition)
Bound volume of 104 pages
Sypha Publications, London, 1998
Ref: ISBN 1 901950 018. Price £19.95.

Richard Chalmers

Architectures for Digital Signal Processing

Architectures for Digital Signal Processing is an interesting technical guide to one of the key areas of modern circuit design. As processors become more powerful and the tasks they must perform become more varied, knowledge of the design of these DSPs becomes more important.

The book aims to bridge the gap between texts covering signal processing algorithms and those covering the implementation and design of microelectronic circuits. There are large quantities of texts covering both these topics and indeed many university courses also make this split. It is a refreshing alternative to have a cross-disciplinary technical text such as this. Originally written by the author in German as a text book, it is geared in its revised and translated form as both a textbook and a reference for those working in the area. The language of the book is easy to follow and logical with certain residual characteristics from the original German text.

The book reviews basic digital circuit design techniques for VLSI (Very Large Scale Integrated) implementation in CMOS (Complimentary Metal-Oxide Semiconductor) technologies. Chapter 3 deals with circuit design techniques

and architectures for implementing basic operations such as addition, subtraction, multiplication and division. General parallel processing and processor pipelining are dealt with in Chapter 4. Methods for mapping algorithms onto array processors are described in Chapter 5. These methods are particularly relevant to adapting architectures to implement regular algorithms.

One of the key applications of DSP design is in the areas of linear filter and discrete Fourier transform implementation, and these are discussed in Chapters 6 and 7. Chapter 8 deals with application-specific programmable signal processors. The characteristics of current DSPs are discussed and it explains methods for increasing the throughput of algorithms in DSPs. Chapter 9 deals with multi-processor systems, while implementation strategies are discussed in Chapter 10. Each Chapter ends with a set of exercises which meet the needs of the textbook reader, but they also aid the reference book reader to understand better the topics treated throughout the book.

Mr Pirsch has written a comprehensive guide dealing with all aspects of Digital Signal Processing architectures and their implementation in circuits. Many of the examples used are drawn from the audio and video processing worlds, but the book is primarily geared towards the student or the professional working in the circuit design area. The philosophy of filling a gap between the disciplines of signal processing algorithm development and the design of the final circuits is interesting, and *Architectures for Digital Signal Processing* provides a useful overview of both areas.

Architectures for Digital Signal Processing
P. Pirsch
Hardbound volume of 419 pages
John Wiley & Sons, Chichester, UK, 1998
Ref: ISBN 0-471-97145-6. Price £39.95

Peter McAvock

Programme Archives

Migration strategies for the digital millennium

EBU
UER

An EBU Seminar on Production Techniques

- *EBU Geneva*
- *26 - 28 January 1999*

This seminar will cover both technical and operational facilities. It should appeal to those who want to develop new archive systems, those who want to exploit the assets in their existing archives, as well as technical and programme executives working in broadcasting and production.

For further information, please contact Mr Jean-Jacques Peters:

Tel: (+41 22) 717 27 21
Fax: (+41 22) 717 27 10
E-mail: peters@ebu.ch

For your diary . . .

EBU meetings, seminars and workshops

January	**Seminar: Programme archives – migration strategies for the digital millennium** *(see page 57)* EBU Geneva, Switzerland 26 - 28 January 1999	**Contact:** Jean-Jacques Peters Tel: +41 22 717 27 21 E-mail: peters@ebu.ch
February	**Technical Committee (13th Meeting)** Geneva, Switzerland 9 - 10 February 1999	**Contact:** Geneviève Juttens Tel: +41 22 717 27 05 E-mail: genevieve.juttens@ebu.ch
May	**Technical Committee (14th Meeting)** Sofia, Bulgaria 3 May 1999	**Contact:** Geneviève Juttens Tel: +41 22 717 27 05 E-mail: genevieve.juttens@ebu.ch
May	**Technical Assembly (4th Meeting)** Sofia, Bulgaria 4 - 5 May 1999	**Contact:** Geneviève Juttens Tel: +41 22 717 27 05 E-mail: genevieve.juttens@ebu.ch

Other meetings, seminars and workshops

January	**4th International DAB Symposium** Singapore 13 - 15 January 1999	**Contact:** Julie Unsworth Tel: +44 171 896 9050 E-mail: unsworth@worlddab.org Web: http://www.worlddab.org/events_frame.htm

International conferences, exhibitions, workshops etc.

April	**NAB 99** Las Vegas, USA 17 - 22 April 1999	**Contact:** NAB Web: http://www.nab.org/Conventions/nab99/default.htm
May	**106th AES Convention** Munich, Germany 8 - 11 May 1999	**Contact:** Martin Woehr Tel: +49 89 590 02434 E-mail: 106th_chairmen@aes.org Web: http://www.aes.org/events/106
June	**Montreux International Television Symposium and Technical Exhibition** Montreux, Switzerland 10 - 15 June 1999	**Contact:** Montreux Symposia Tel: +41 21 963 32 20 E-mail: message@symposia.ch Web: http://www.montreux.ch/symposia/
July	**SMPTE 99 Conference and Exhibition** Sydney, Australia 13 - 16 July 1999	**Contact:** Expertise Events Pty Tel: +61 2 9977 0888 E-mail: smptepapers@bigpond.com Web: http://www.smpte.org/conf/smpte99.html
September	**IBC 99** Amsterdam, The Netherlands 10 - 14 September 1999	**Contact:** IBC Office, London Tel: +44 171 240 3839 E-mail: show@ibc.org.uk Web: http://www.ibc.org.uk/ibc

EBU Active Members

(last update: November 1998)

Algeria
Entreprise Nationale de Télévision / Entreprise Nationale de Radiodiffusion Sonore / Télédiffusion d'Algérie

Austria
Österreichischer Rundfunk

Belarus
Belaruskaja Tele-Radio Campanija

Belgium
Vlaamse Radio en Televisie and Radio-Télévision Belge de la Communauté française

Bosnia-Herzegovina
Radio Televizija Bosne i Hercegovine

Bulgaria
Bâlgarsko Nationalno Radio
Bâlgarska NationalnaTelevizija

Croatia
Hrvatska Radiotelevizija

Cyprus
Cyprus Broadcasting Corporation

Czech Republic
Cesky Rozhlas
Ceská Televize

Denmark
Danmarks Radio
TV2/Danmark

Egypt
Egyptian Radio and Television Union

Estonia
Eesti Raadio
Eesti Televisioon

Finland
MTV Oy
Oy Yleisradio Ab

France
Grouping of French broadcasters, comprising:
- Télévision Française 1
- France 2
- France 3
- Canal Plus
- Radio France
- Radio France Internationale
- TéléDiffusion de France

Europe 1

Germany
Arbeitsgemeinschaft der öffentlich-rechtlichen Rundfunkanstalten der Bundesrepublik Deutschland (ARD), comprising:
- Bayerischer Rundfunk
- Hessischer Rundfunk
- Mitteldeutscher Rundfunk
- Norddeutscher Rundfunk
- Östdeutscher Rundfunk Brandenburg
- Radio Bremen
- Saarländischer Rundfunk
- Sender Freies Berlin
- Südwestrundfunk
- Westdeutscher Rundfunk
- Deutsche Welle
- DeutschlandRadio

Zweites Deutsches Fernsehen

Greece
Elliniki Radiophonia – Tileorassi SA

Hungary
Magyar Rádió
Magyar Televízió

Iceland
Ríkisútvarpid

Ireland
Radio Telefís Éireann

Israel
Israel Broadcasting Authority

Italy
RAI–Radiotelevisione Italiana

Jordan
Jordan Radio and Television Corporation

Latvia
Latvijas Valsts Televizija
Latvijas Radio

Lebanon
Radio Liban / Télé-Liban

Libya
Libyan Jamahiriya Broadcasting

Lithuania
Lietuvos Radijas ir Televizija

Luxembourg
CLT Multi Media
Etablissement de Radiodiffusion Socioculturelle du Grand-Duché de Luxembourg

Former Yugoslav Republic of Macedonia
MKRTV

Malta
Broadcasting Authority – Malta / Public Broadcasting Services Ltd – Malta

Moldova
Teleradio-Moldova

Monaco
Groupement de Radiodiffuseurs monégasques, comprising:
- Radio Monte-Carlo
- Télé Monte-Carlo
- Monte-Carlo Radiodiffusion

Morocco
Radiodiffusion-Télévision Marocaine

Netherlands
Nederlandse Omroep Stichting (NOS), comprising:
- Algemene Omroepvereniging AVRO
- Vereniging de Evangelische Omroep
- Katholieke Radio Omroep
- Nederlandse Christelijke Radio Vereniging
- Nederlandse Programma Stichting
- Omroepvereniging VARA
- Omroepvereniging VPRO
- TROS

Norway
Norsk rikskringkasting
TV 2 AS

Poland
Polskie Radio i Telewizja, comprising:
- Telewizja Polska SA
- Polskie Radio SA

Portugal
Radiodifusão Portuguesa SA
Radiotelevisão Portuguesa SA

Romania
Societatea Româna de Radiodifuziune
Societatea Româna de Televiziune

Russian Federation
Obshchtestvennoe Rossijskoe Televidenie
Radio Dom Ostankino, comprising:
- Radio Mayak
- Radio Orpheus
- Voice of Russia

Rossijskoe Televidenie

San Marino
San Marino RTV

Slovakia
Slovensky Rozlas
Slovenská Televizia

Slovenia
Radiotelevizija Slovenija

Spain
Radio Popular SA COPE
Radiotelevisión Española
Sociedad Española de Radiodifusión

Sweden
Sveriges Television och Radio Grupp, comprising:
- Sveriges Television Ab
- Sveriges Radio Ab
- Sveriges Utbildningsradio Ab

Switzerland
Société Suisse de Radiodiffusion et Télévision

Tunisia
Établissement de la Radiodiffusion-Télévision Tunisienne

Turkey
Türkiye Radyo – Televizyon Kurumu

Ukraine
Natsionalna Radiokompanya Ukraïny
Natsionalna Telekompanya Ukraïny

United Kingdom
British Broadcasting Corporation
United Kingdom Independent Broadcasting, comprising:
Independent Television: The Network Centre, grouping :
- Anglia Television
- Border Television
- Carlton Television
- Central Independent Television
- Channel Television
- Grampian Television
- Granada Television
- HTV
- London Weekend Television
- Meridian Broadcasting
- Scottish Television
- Tyne Tees Television
- Ulster Television
- Westcountry Television
- Yorkshire Television
- Independent Television News

Channel 4, Sianel 4 Cymru
Commercial Radio Companies Association

Vatican State
Radio Vaticana

EBU Associate Members

(last update: November 1998)

Albania
Radiotelevisione Shqiptar

Armenia
Hayastani Azgayin Radio and Hayastani Azgayin Heroustatesoutun

Australia
Australian Broadcasting Corporation
Federation of Australian Commercial Television Stations
Special Broadcasting Service

Bangladesh
National Broadcasting Authority of Bangladesh

Barbados
Caribbean Broadcasting Corporation

Brazil
TV Globo Ltda

Canada
Canadian Broadcasting Corporation

Chile
Corporación de Televisión de la Universidad Católica de Chile
(Canal 13)

Cuba
Instituto Cubano de Radio y Televisión

Greenland
Kalaalit Nunaata Radioa

Hong Kong
Asia Television Ltd
Radio Television Hong Kong
Television Broadcasts Ltd

India
All India Radio

Iran
Islamic Republic of Iran Broadcasting

Japan
Asahi National Broadcasting Co. Ltd
(TV Asahi)
Fuji Television Network Inc
National Association of Commercial Broadcasters in Japan
Nippon Hoso Kyokai
Nippon Television Network Corporation
Tokyo Broadcasting System Inc
Tokyo FM Broadcasting Co. Ltd

Korea (Republic of)
Korean Broadcasting System
Munhwa Broadcasting Corporation

Malawi
Malawi Broadcasting Corporation

Malaysia
Radio Television Malaysia

Mauritius
Mauritius Broadcasting Corporation

Mexico
Televisa SA de CV

Nepal
Nepal Television Corporation

New Zealand
Radio New Zealand
Television New Zealand Ltd

Oman
Oman Directorate General of Radio and Television

Pakistan
Pakistan Television Corporation

South Africa
South African Broadcasting Corporation

Sri Lanka
Sri Lanka Broadcasting Corporation

Syria
Organisme de la Radio-Télévision Arabe Syrienne

United Arab Emirates
Emirates Broadcasting Corporation
United Arab Emirates Radio and Television – Dubai

United States
Capital Cities / American Broadcasting Companies Inc
CBS Inc
Corporation for Public Broadcasting/Public Broadcasting Service / National Public Radio / Public Radio International
National Broadcasting Company Inc
Turner Broadcasting System Inc
United States Information Agency
WFMT

Venezuela
Corporación Venezolana de Televisión CA
Radio Caracas Televisión / Radio Caracas Radio

Zimbabwe
Zimbabwe Broadcasting Corporation

Approved Participants
Antenna Hungária
ARTE
Euronews
Israeli Educational Television
JP "MRD"
Middle East Broadcasting Centre Ltd
Sentech (Pty) Ltd
TV5